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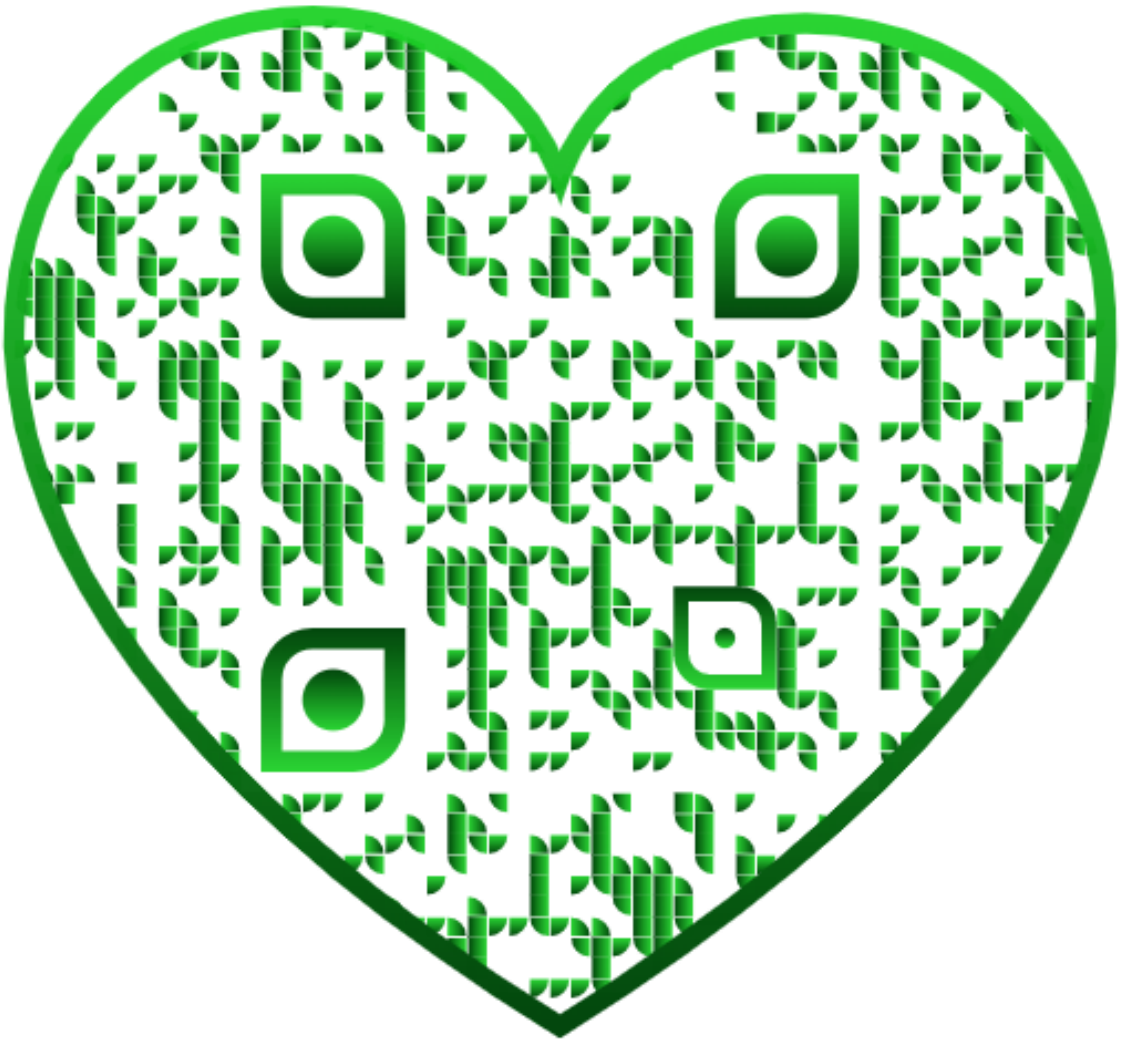

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Guitarist Justus Hajas, Photo by Meg Bauman

WORLDS APART

This month, I've reviewed not one but two moving-coil dynamic microphones. In some ways, they have much in common. Both are intended for close-up use, and both have special features designed to control proximity effect. Yet they are also polar opposites. Warm Audio's WA-19 is a recreation of a classic mic from the '60s, with a capsule "meticulously assembled by hand". Shure's Nexadyne is right at the cutting edge, with a dual-capsule assembly so complex that it couldn't be assembled by hand.

The contrast captures in microcosm the difference between the contemporary studio and live sound worlds. The WA-19 will be a popular choice for recording, while the Nexadyne will be ubiquitous on stages everywhere. But why isn't it the other way around?

Marketing decisions obviously play a role, but there's more to it than that. Assuming the Nexadyne really does offer the advantages Shure claim, those will have tangible results on stage: more clarity, more gain before feedback, a more controlled vocal tone and less need to compensate for poor mic technique. If it's technically the best tool for the job, live sound engineers will be hard-headed enough to recognise that and buy into it.

In the studio, by contrast, we rarely push equipment to its technical limits. Few of us record vocals live in the room with the band any more, and when the singer is overdubbed after the fact, the off-axis properties of the mic become much less important. Studio engineers have time, and choice. We can put up multiple mics, or swap them out when they don't suit. And we are free to apply EQ, compression, saturation and other processing without unintended consequences. Ultimately, we can enjoy the characterful qualities of vintage-style gear in the studio because we don't need to worry about its limitations — and that's a luxury live sound engineers don't have.

Whether you're a live sound or studio person, however, there will be plenty to interest you at *Sound On Sound*'s very own gear showcase. GearFest UK is back at Tileyard in London's King's Cross on Saturday July 13th, with over 100 brands already confirmed as exhibitors and an unmissable programme of talks. It's your chance to get hands-on both with vintage kit and the latest cutting-edge technology, and to talk to the people who make it. Get your tickets now at www.gearfestuk.com — and be sure to turn up early on the day, because the first 100 people through the doors will be entered into our massive prize draw. You could be leaving GearFest with a lot more classic and modern gear than you had when you went in...

Sam Inglis
Editor In Chief

"Ultimately, we can enjoy the characterful qualities of vintage-style gear in the studio because we don't need to worry about its limitations."

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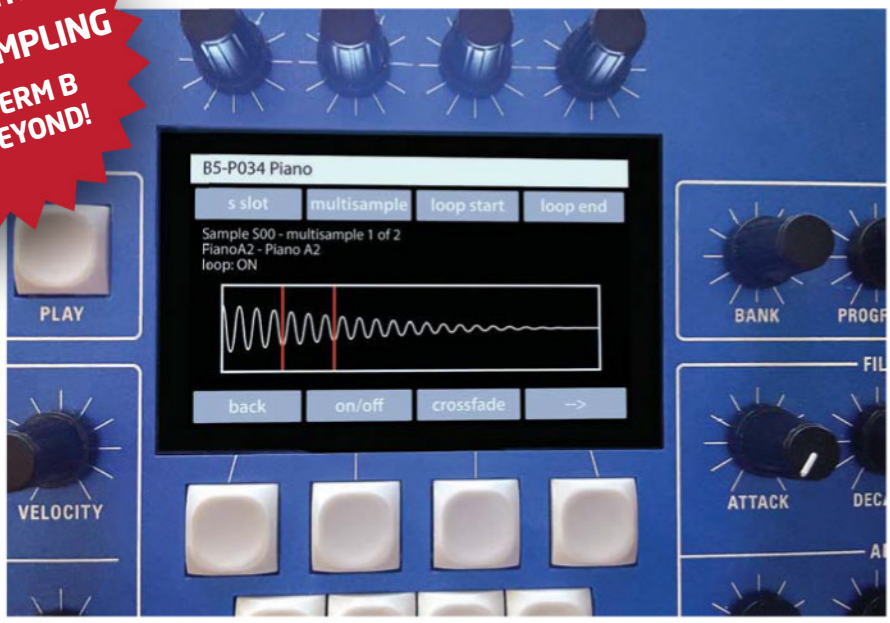

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Pearl ELM-T

Valve Microphone

The ELM-T is a new microphone that has been 70 years in the making.

SAM INGLIS

If you look up "company man" in an English-to-Swedish dictionary, you'll probably see a photo of Bernt Malmqvist. Since joining in 1954 as a machinist, he's spent his entire career with Pearl Microphones and sister company Milab. Following in the footsteps of company founder Rune Rosander, Bernt has been responsible for many innovative and successful designs, and on his formal retirement in 2017, the Pearl and Milab brands were reunited under one roof.

After 63 years designing, building and selling microphones, many people would have seen retirement as an opportunity to do something else. Not Bernt Malmqvist, who decided to add to his already considerable legacy by developing the ultimate valve microphone. Several years in the making, the ELM-T Bernt Malmqvist Signature Edition is a no-compromise design that draws on all of its creator's vast experience.

Straight & Narrow

One of the core technologies at Pearl and Milab is the rectangular capsule. Pioneered by Rune Rosander back in the 1950s, this has some practical advantages over the conventional circular design.

"What's striking about the vocal capture that you get from the ELM-T is that there's no hint of hardness."

The tensioned diaphragm does not have a single, dominant resonant frequency like its circular counterpart, and the same surface area can be achieved using a smaller diameter, allowing some Pearl and Milab models to offer large-diaphragm performance with the footprint of a small-diaphragm mic. The rectangular capsule also exhibits different pickup patterns in the vertical and horizontal planes, which can be a useful quality in real-world recording scenarios. For example, if you're forced to work in a room with a low ceiling, a mic that has much stronger rejection along the vertical axis can be a valuable asset.

Designed by Bernt Malmqvist and introduced in the mid-2000s, Pearl's ELM capsule took the rectangular design, quite literally, to new heights. Previous Pearl and Milab capsules had typically employed a 2:1 aspect ratio, but the ELM capsule dramatically ups that to 7:1. In doing so, it heightens the contrast between vertical and horizontal polar responses. The ELM-C is a fixed-cardioid mic, the -B is a figure-8 model, and the -A allows the outputs from both sides of the capsule to be recorded independently, meaning that the polar pattern can freely be varied after the fact.

The new ELM-T is closest in spirit to the ELM-A, offering a polar pattern that is continuously variable from figure-8 at »

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» one extreme to omni at the other, with cardioid response in the middle. Unlike the ELM-A, though, this is set at the time of recording, using a rotary knob on the power supply. Said PSU is a third-party model made by Andreas Grosser, and appears to be of very high quality. Apart from the pattern control and on/off switch, it features a standard IEC mains inlet, an equally standard XLR output and a single multi-pin connector for the supplied custom cable; there's no pad or filter.

The mic itself is, as you might expect, unusually long and slender for a valve model, measuring 238 x 42mm, and connects to the cable using a small-format Lemo connector rather than the more usual screw-on Binder type. It's finished in a lovely "Rolex green", and as always with Pearl and Milab mics, the build quality is excellent. A Rycote USM shockmount is supplied; in my experience these aren't always the best choice for very long and thin mics, since they only grip at one point along the body, but I encountered no problems with the review model. Mic, PSU, mount and cables are housed in a smart and robust flightcase.

Pearl don't state what valve is used in the ELM-T, and it's not visible from the outside, but the specs are fairly typical of a good modern design. Sensitivity is a comfortable 12mV/Pa, and self-noise a very respectable 16dBA, while 1% THD is reached at 126dB SPL. The published frequency response chart is almost ruler-flat in cardioid, with slight deviations creeping in towards the very top of the spectrum in omni.

The Pure Path

These days, we often think of valve mics as a means of introducing character into a recording. But that wasn't the intention of the designers behind the classic Neumann and AKG models, nor that of Rune Rosander when he developed the first generation of Pearl valve mics back in the 1950s. At that time, valve technology was simply the only way to approach the universal goal of capturing a source sound as faithfully as possible.

Bernt Malmqvist is one of the last surviving links to the golden age of mic design, so unsurprisingly, he hasn't set out to give the ELM-T a deliberately coloured sound. If you approach it in the hope of adding obvious saturation, rock & roll grit or "tube warmth" to your recordings, you're likely to be underwhelmed, because what the

The ELM capsule is seven times as long as it is wide, giving it unusual directional characteristics.

ELM-T offers is a beautifully clean, balanced, tonally neutral sound. There's none of the midrange push that characterises some classic Neumann designs, nor the slightly hyped high-frequency sparkle you might expect from a C12-type mic. The low-mid richness that we sometimes associate with valves and transformers is also conspicuously absent. The magic of smoothing means that a flat published frequency response can conceal all sorts of idiosyncrasies in the detail, but in this case, it's entirely believable.

Yet, at the same time, the ELM-T certainly isn't boring. It's one of very few mics I've tried that seems equally at home with male and female vocalists, regardless of register and singing style. I don't know whether it's the capsule or the valve that is responsible, but what's striking about the vocal capture that you get from the ELM-T is that there's no hint of hardness: the sound always remains perfectly smooth, without being noticeably dark or too soft to cut through the mix. Sibilants are neither exaggerated nor 'smooshed', and the ELM-T won't highlight negative qualities such as honkiness or nasality. I've never tried any of the solid-state ELM models, but my sense is that what the valve circuitry brings to the party is a forgiving quality that enables it to swallow up harshness and retain its poise no matter what is thrown at it.

My feelings about the ELM-T in many ways mirror the reaction I had to Neumann's reissue of the M49 a couple of years ago, and indeed the two mics do have some things in common, notably the continuously variable pattern control. In both cases, sensitivity is somewhat pattern-dependent, and appears to increase in figure-8. However, whereas the tonality of the M49V is also quite pattern-dependent — and, to my ears, rather less attractive in figure-8 — the ELM-T is impressively consistent in this respect. There is, naturally, more proximity effect in bidirectional mode, and the ratio of direct to reflected sound varies as you'd expect, but you never lose that supremely balanced, natural sound. With its high-frequency response flat to 20kHz and beyond, the ELM-T doesn't sound at all 'vintage', as the M49 might, but it gives you the same sense of pure, understated class — and the confidence that it will able to capture absolutely anything. If Rune Rusander, or any of the other great designers from the 1950s, had had the means to come up with the ELM-T, I'm sure they would have jumped at the chance. And if Bernt Malmqvist finally does decide to bow out of the microphone game, he can be justly proud to do so with the ELM-T.

summary

The ELM-T distils seven decades of design experience into a true Rolls Royce of a mic.

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Sonnect SoundWire

In-line USB D-A Converter

This clever cable is as smart as it looks!

HUGH ROBJOHNS

Back in February 2022, I reviewed a superb little audio tester from a start-up company called Sonnect. Based in Italy, Sonnect are the brainchild of David Scorteccia, a professional live-sound engineer working in the UK. I was very impressed with the design and engineering of the Sound Bullet (www.soundonsound.com/reviews/sonnect-audio-sound-bullet) and I remember talking at length with David about other potential products at the time — so I was very pleased recently to be asked to evaluate the company's second product, the SoundWire.

Nearly three years in development, the SoundWire is equally innovative as the Sound Bullet in solving a perennial problem very neatly and elegantly. In essence, it's a high-quality, two-channel, plug-and-play USB-C audio interface/D-A converter, providing a stereo audio output from laptops, tablets and smartphones. So far, so normal — but this product isn't a bulky box, doesn't need external powering, provides balanced line-level audio on XLRs, and fully protects the source against accidental application of phantom power. In fact, it looks like nothing more than a simple adaptor cable, completely disguising the sophisticated electronics within.

The Problem

I'm sure we've all been in the situation of needing to get audio out of a laptop and into a mixing console. The usual options are either to take the output from the laptop's built-in soundcard via a 3.5mm mini-jack in the headphone socket, or derive digital audio via a USB port. The analogue route is simple but often suffers low-quality audio, intermittent connection problems, and potential grounding issues. And, while a direct connection from laptop to mixing console is possible using a mini-jack to dual-XLR 'bodge cable', it only needs the accidental activation of phantom power on those mixer channels to destroy the laptop's audio circuitry — which really isn't good for client satisfaction!

The USB-C connector is slightly longer than most, to accommodate the (up to 192kHz) D-A converter.

So, to be safe, it's essential to connect the laptop via some form of isolation transformer, and that means either a stereo DI box or a two-channel line-isolation box. This guarantees phantom-power protection for the laptop (as well as, conveniently, balanced audio connections back to the console), but it needs another box or two and more cables and adaptors, so it soon starts to get a bit bulky, cumbersome and expensive.

Alternatively, using a USB output needs some form of dedicated USB audio interface to perform the required D-A conversion. That usually means another bulky, expensive box, and most USB interfaces, at least for Windows devices, expect specific audio drivers to be installed. Moreover, it remains necessary to use DI or line-isolation boxes to provide phantom-power protection, this time for the USB interface itself.

Either way, it's always a complete faff to set up, with lots of adaptor cables, bulky interface boxes, and potentially multiple points of failure and confusion. The SoundWire addresses all of these issues with an incredibly simple and totally straightforward solution, which is no more difficult than plugging in a cable.

The SoundWire Solution

The SoundWire cable itself weighs just 140g and measures roughly 2.8m overall: 2.3m from the USB-C connector to a plastic Y-splitter joint, with the separate XLR output tails adding a further 0.5m. In other words, it's more than long enough to run from a laptop placed alongside a console directly into its rear-panel inputs, or a presentation podium down to a stagebox on the floor, without needing XLR extensions. One cable, three plugs: job done!

While USB-C is the modern standard, there are a lot of legacy USB-A ports still out there, so Sonnect offer a compact USB-A adaptor as an optional accessory, and I'd highly recommend acquiring one for use with the SoundWire. The USB-C plug (which has gold-plated pins) slots neatly into this adaptor, forming a robust and reliable connection and looking much like an ordinary, if slightly chunky, USB-A plug. The USB-C connector itself is a little longer than usual, and that's because inside is a teeny-tiny surface-mount D-A converter (an ESS 9272 Sabre DAC), along with all the associated support circuitry. This is directly bus-powered, obviously, drawing just 100mA, and the converter supports all sample rates up to 192kHz, with word lengths up to 32 bits. It's also fully class-compliant so doesn't need any drivers, regardless of the computer host. It is literally a case of plugging in, letting the computer recognise it, selecting it as the computer's desired audio output destination, and that's it.

At the opposite end to the USB-C plug is a pair of Neutrik XLRs, again with gold-plated pins. There's technology »

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» hidden here, too: each XLR contains a custom, miniaturised transformer to balance the output and prevent phantom power reaching the electronics. The XLRs are clearly identified as L and R on a blue ring between the shell and strain relief.

The wiring linking the three plugs (and their electronics) is protected within a tough, 4.5mm, blue, woven-nylon sleeve, and it is clear that this product has been designed by somebody with real-world experience, because an elastic toggle is permanently attached to the Y-splitter, and this wraps around the coiled-up cable and clips into a slot to keep it tidy and tangle-free.

An optional storage pouch, slightly larger than a CD case, is also available, with a magnetic closure, made of a tough nylon at the rear and a neoprene material on the front, and an internal loop is included to store the optional USB-A adaptor. All three elements of the SoundWire system look very well made and capable of surviving a long life on the road.

The line-level analogue outputs are on XLR plugs, and the device is protected from the application of phantom power from a mixer or preamp.

Technicalities

The SoundWire delivers a nominal 0dBu balanced analogue output if presented with a full-level 0dBFS digital source. This choice of audio level might seem a little odd, but it's ideal for live-sound applications since it works acceptably whether connected to a mic preamp (set with low gain and/or a pad) or a line input (with some added gain). It's roughly the same as might be expected from a modern keyboard, for example.

On the test bench the review sample actually delivered +0.17dBu on both channels (matched within 0.01dB) for a 0dBFS output, and the frequency response measured flat within 0.2dB across the entire 20Hz to 20kHz audio bandwidth. I noticed a very slight tilt downwards from low frequencies to high frequencies (0.15dB), which is obviously inaudible, but I wondered if this was affected by the load impedance, as transformer outputs often are. The above figures were obtained with a high-impedance (>20kΩ) destination, and switching to a very low, legacy 600Ω load impedance did change things. The maximum output level reduced to -1.7dBu, and the frequency response tilt changed to a gentle upwards slope, enhancing high frequencies slightly. These variations are entirely typical and are insignificant in practical terms — to all intents and purposes the SoundWire performs consistently regardless of the likely load impedances encountered in normal applications.

Running the AES17 Dynamic Range test, I achieved a figure of 110.5dB (A-weighted). That's slightly better than the published specifications but on the low side by modern standards, where 116dB or higher is fairly common in modern interfaces and converters. My rule of thumb is that anything better than 116dB is of professional mastering quality, and anything below 100dB is junk. But while the SoundWire is 6dB below mastering grade, it's far from being poor (it's still 6dB better than Behringer's ADA8200, for example, with which many users are perfectly happy) and, without any doubt, it's the best-performing USB adaptor cable I've ever measured! The way I look at it, the SoundWire prioritises convenience and practicality slightly more than mastering sound quality, which is entirely reasonable, yet still manages to achieve a perfectly decent audio quality that's ideally suited to live-sound applications.

One minor oddity arose during my bench tests, but Sonnect identified it before I did: the review unit's output signal polarity was inverted on both channels. I only do bench tests as I'm writing the review but had been using the SoundWire extensively before that; absolute signal polarity is notoriously hard to identify in normal auditioning with random audio sources. I'm told that this error affects only early production units and that there's a simple fix via a firmware update, which should be available by the time you read this. Early adopters can update their SoundWires via the firmware update by contacting Sonnect directly, but really this foible is irrelevant in typical live-sound applications, so will have been completely unnoticed by most early adopters (as it was by me). It's also easily corrected by the desk channel's polarity inversion button. Of course, later production units will be corrected during manufacturing.

At present, the drivers are generic and while the latency performance is fine for playback applications, the delay becomes noticeable if monitoring live-played virtual instruments. However, it can be used with third-party drivers like ASIO4ALL, and Sonnect tell me they're developing dedicated drivers for both Mac and Windows which will make it better suited in scenarios where very low latencies are desirable.

Conclusion

The SoundWire's simple form hides a lot of complex, bespoke technology that pushes up the price (on a par with budget bidirectional USB interfaces). But then it completely obviates the need for stereo DI or transformer-isolation boxes, is fully class-compliant for plug-and-play usability, removes all safety concerns about phantom-power catastrophes, and provides transformer-balanced output signals at a convenient level. And it sounds very good — it's great that the focus on practical convenience hasn't sacrificed the audio quality. All of which should make the SoundWire very attractive for anyone who regularly connects laptops and similar devices to a mixer. In fact, having used it on several occasions now, I'm very reluctant to go back to my adaptor cables, DI boxes, or traditional laptop interfaces.

summary

A well-designed, well-engineered, elegant and highly practical means of delivering high-quality balanced analogue audio from any laptop or tablet.

$ SoundWire €122 (about $125). Storage pouch €25 ($26). USB-A to USB-C adapter €10 ($11).
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WILLIAM STOKES

Endorphin.es Golden Master

Multiband Processor Pedal

Designed to deliver an aggressive ’90s mastering sound on stage, the Golden Master could well come in handy in the studio too.

Alongside the Ghost Pedal that I reviewed — and was impressed by — last month, Endorphin.es released the Golden Master Pedal, and I think this one pushes the guitar pedal format about as far as it could possibly go into studio outboard territory without physically morphing into a 19-inch rack unit! It’s easy to see why these two pedals were announced within a day of each other too: although they’re both impressive on their own, they form something of a powerhouse when paired together.

Design & Functionality

Like the Ghost Pedal, the Golden Master Pedal is essentially a 24-bit/96kHz adaptation of a Eurorack module of the same name, but it benefits from a few welcome enhancements and additions as well as a little extra panel space (the Eurorack module crammed the Golden Master’s functionality into just 6HP, which was no mean feat!). It’s a stereo multiband processor that’s intended as a quick and intuitive end-of-chain device for delivering ‘high pressure audio’, particularly in a live setting. To that end, it offers EQ, compression and Mid-Sides processing in three frequency bands, and while it simplifies some aspects of this arrangement others remain highly tweakable.

Endorphin.es cite their love for the distinctive character of 1990s mastering techniques as a prime motivation behind the Golden Master’s design, as well as extensive experience with live music performance. I can believe it too — if you haven’t listened to the music of artist and company co-founder Julia Bondar for yourself, I strongly encourage you to do so. As for the 1990s mastering reference, you’d do well to bear in mind the kind of no-prisoners-taken loudness of music by the likes of the Prodigy or Underworld. This is not to say it’s a case of ‘off or on’, but the Golden Master is capable of endowing your signal with all the angular, pumping aggression you could want on a master bus.

Considering what it’s possible to achieve with the Golden Master, its control panel is surprisingly modest. There are knobs for its three bands (labelled Low, Mid and High) along with three respective mute buttons. Above these are controls for volume, compression threshold and up to +10dB of input gain — that’s enough to push a signal into good-sounding saturation that, to my ears, remains blissfully free of digital artefacts. There’s also a lovely looking and nicely responsive stereo level meter.

At the bottom of the panel are two footswitches, one for bypassing the pedal (with true or buffered bypass modes usefully on offer) and one for switching between the three available presets. On the left is a Mode button, with LED indicators for each of the three modes, and another button for switching the meter mode between input (after gain) and output. Needless to say, the Ghost and Golden Master pedals are cut from the same cloth in terms of architecture and construction. For instance, they employ the same mechanism for saving and loading presets. One conspicuous difference is that while the Ghost offers nine preset slots, the Golden Master has just three. It’s probably the right call, though, given the desire to keep things simple — it’s unlikely you’ll really need more than that for an end-of-chain processor.

The rear panel hosts the stereo inputs and outputs, as well as an expression pedal input. I was excited to discover that the Golden Master serves as a capable DI box: its pseudo-balanced outputs cover a multitude of sins when it comes to interfacing with a PA system and maintaining control when playing electronic instruments live. (A brace of TRS jack to XLR cables will likely be the next thing on your shopping list!) The expression pedal input also serves as a side-chain input, with only a mildly confusing front-panel button combo required to enter the mode in which you set the bands and parameters this affects. »

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The expression pedal socket doubles up as an external side-chain input for the multiband compressor.

» The Mode button has an arrow pointing up towards the mode LEDs. From bottom to top, these are EQ, compression and Mid-Sides, and this suits the typical workflow: I found that whether used per-band or holistically, I generally preferred to start by adjusting the EQ, then hitting the Mode button to move to compression, before hitting it once more to work on the Mid-Sides balance. Rinse and repeat. Of course, navigating knob positions presented a hurdle because, with all that mode switching, it's rare that knob values match their parameters' actual values. As it is, parameters snap to their represented values as soon as the knobs are turned. Again, all things considered this is probably the right choice as it avoids confusion, even if it does occasionally risk clunkiness.

The three EQ bands (20-300 Hz, 300Hz to 3kHz and 3-20 kHz) offer -20dB to +8dB of gain, which is similar to the response of typical DJ mixers. This also suggests that when balancing your image with the Golden Master, one is encouraged to start by attenuating frequencies rather than boosting them. This said, it's not like the tonal response of positive values is unsatisfactory. Far from it, in fact: boosting the highs just the right amount often resulted in some crisp upper-end enhancement, while carefully working the lows added pleasant beef without too much murkiness.

Compression Made Easy

When it comes to compression, I can imagine a fairly strenuous R&D process was needed to arrive at the Golden Master's final design. Controls for typical compressors (at the most basic level, the threshold, ratio and make-up gain) would quickly have congested and confused the layout. Instead, then, some parameters here are adjustable while others are fixed. Each band knob controls all of its primary compression settings (attack, release, ratio and threshold) one-knob style, with anticlockwise values starting at 50ms attack, 1500ms release, 1:1.25 ratio and -12dB threshold. Turn the knob fully clockwise, and those values reach a maximum of 14ms attack (faster), 15ms release (also faster), 1:4 ratio (higher) and -20dB threshold (lower).

These controls might appear a little limiting, and they would be if it weren't for the additional ±10dB Global Threshold knob at the top. This makes it possible to adjust compression across the three bands to taste, and then bring the threshold for the whole lot down or up and thus change the response of the multiband component. In practice, it's something more like having two layers of one-knob compressors to play with than tweaking an actual outboard compressor, and this suits the Golden Master's self-described aptitude for live performance perfectly. After all, no-one wants to be faced with a compression knee control when they're simply trying to get their bass pumping louder on stage. There's also no real risk of things getting really out of hand signal-wise, since the very end of the Golden Master's signal chain has a brickwall limiter, and there's also a fixed 1.5ms look-ahead to avoid the clicky distortion of ultra-fast attack times. So it really is a case of permission granted for wild knob tweaking.

"The Golden Master is capable of endowing your signal with all the angular, pumping aggression you could want on a master bus."

Finally The Mid-Sides mode does what you'd imagine for each band, supposing you're feeding it a stereo image, of course. When centred, each band's knob is in its default position. Turned fully anticlockwise it attenuates the Sides component, and when fully clockwise it attenuates the Mid signal. There's the option to adjust this with the lowest band too — it's rarely advisable to have the lowest frequencies of a sound or a mix anywhere but the Mid channel, but hey, who are Endorphin.es to tell us that?! In any case, it was hugely satisfying hearing the difference when widening the image the further up the spectrum, as it could often add a real sense of three-dimensionality to the overall sound.

Sound & Control

EQ, multiband compression and multiband Mid-Sides processing are useful functions in and of themselves, and it's rare to find some of them in a pedal, and certainly all in one. So it's already a little different and potentially useful. But the Golden Master is so special in the way it combines the three to create a fast and intuitive workflow, and that makes it well suited to use during a performance — just a couple of twists and button pushes can both tighten and centre your low end while widening your mid frequencies and enhancing your highs.

In terms of its sound, the Golden Master punches well above its weight. In fact, there wasn't anything I fed through it that didn't sound noticeably better for it. In this regard, I can actually foresee it having to battle the inevitable preconceptions that will be triggered by its guitar pedal form factor. Add its DI box functionality, and you've got a truly indispensable tool, units of which I can imagine littering the stage for any live electronic act. In fact, I might well have to get my hands on a few more!

summary

A high-quality, great-sounding multiband processor that's intended for use at the end of any chain, the Golden Master has a well conceived control set that makes it perfect for use during a live performance.

$ €295 (about $300).
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Sonible pure:unmask

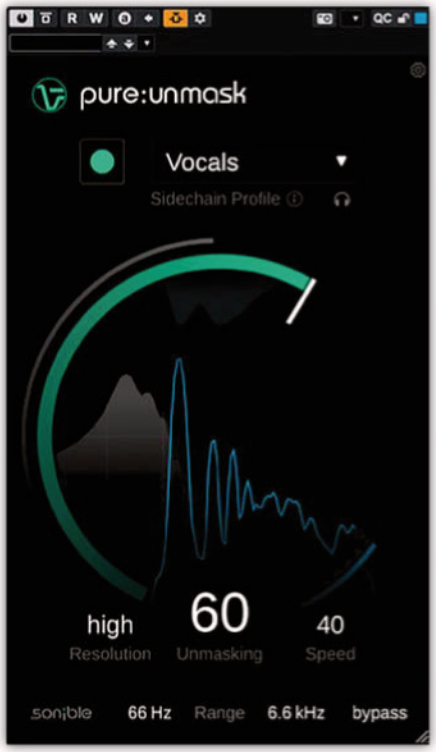
Spectral Ducking Plug-in

Can it really be this quick and easy to tackle masking in your mix?

JOHN WALDEN

Regular readers will know that I reviewed Sonible's pure:EQ plug-in in *SOS* December 2023 and was really impressed. Sonible's 'pure' concept takes elements of the AI-based processing found in their flagship 'smart' series of plug-ins and distils things down into a much simpler, easy-to-use format. In fact, the user experience is almost fool-proof, and relatively routine but usually time-consuming tasks such as EQ (pure:EQ), compression (pure:comp), limiting (pure:limit) and reverb (pure:verb) can be dealt with speedily and effectively. They've now added pure:unmask to this series. A similarly streamlined plug-in, this one's primary function is to tackle the problem of elements in your mix being unhelpfully masked by others.

Masking is a phenomenon that occurs when two audio sources contain overlapping frequency content, and the louder one obscures the other in that part of the spectrum. It can lead to an inconsistency in tone, or sometimes to sounds seeming to drop out of a mix altogether. Ideally, you'd find a way to tuck one sound out of the way of the other, so that both can do what the song requires. For example, you might want a bass guitar to dip out the way of a kick drum, or a bed of guitars and keyboards to make way for a lead vocal. While you can do this with EQ to some extent, a more sophisticated tactic is to route a signal from the high-priority track (such as a lead vocal) into the side-chain input of a compressor or dynamic EQ on the lower-priority track, so the latter only makes way when both instruments are playing.



Listen & Learn

pure:unmask attempts to automate the second of these options. Once installed as an insert effect on the lower-priority track, and with the side-chain routing configured from your higher-priority track (a fairly straightforward process in most DAWs now), pure:unmask applies its AI-based decision-making to configure some spectral processing which, in essence, produces results akin to dynamic EQ, with overlapping frequencies in the lower-priority track being ducked, ensuring that the main event retains its clarity, and that there's minimum 'damage' to the lower-priority track when there's nothing going on in the other one.

> "pure:unmask is a simple and effective problem solver: in practice it just works."

Having set up the side-chain routing, the large green button is used to trigger an initial 'listen and learn' process, through which the plug-in evaluates a short section of both audio signals. You can give this process a nudge in the right direction by selecting a profile type from the drop-down list — this identifies the nature of the high-priority signal and has profiles for vocals, speech, guitar, bass, drums, kick and Universal, which is a catch-all for other audio types.

Once the learning's done, pure:unmask configures itself and, fingers crossed, on playback you should hear that extra bit of clarity in your higher-priority sound. The central portion of the GUI indicates what's happening, with the spectrum of both tracks being displayed along with the spectral ducking being applied, and the changes to the lower-priority track as a result.

That may well be 'job done', but you have some options to finesse the result if needed. Most importantly, you can adjust the amount of unmasking being applied, but you also can define the resolution of the spectral processing, the frequency range over which it's applied, and the attack/release times — all useful options that don't overcomplicate things.

Let It Shine!

Like all the pure series plug-ins, pure:unmask is a simple and effective problem solver: in practice it just works. If you have a really complex combination of audio signals that need their tonal balances untangling, you'll perhaps want a more sophisticated toolset, but for pretty much all routine unmasking jobs this is an excellent time-saving tool. The price is sensible too, and if you manage to pick up the whole pure bundle in one of Sonible's sale events I'd say it's an absolute steal. ■■■

summary

A great addition to Sonible's pure series of plug-ins, pure:unmask makes it easy to claw back some clarity in your mix.

$ $49 (discounted to $29 when going to press).
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Toontrack Session Organ EKX

EZkeys Expansion

Toontrack bring the world's favourite organ to EZkeys 2.

JOHN WALDEN

Toontrack's EZkeys 2 was an undoubted triumph when released just over a year ago (see the August 2023 issue for a full review). A number of EKX expansions have appeared since then, providing additional piano-based sound choices (the excellent Soul Roads EKX, for example, which we reviewed in the September 2023 issue) and there have also been non-piano options offered such as the Synthwave EKX with its cool collection of '80s-inspired synth sounds.

However, a popular request from within the EZkeys user community has been for a Hammond B3 expansion. This is an instrument with a special place in music history and its iconic sound can be found in a wide range of genres, from its origins as a replacement for the church pipe organ to its adoption in gospel, jazz, blues, classic rock, prog and pop.

Toontrack have now answered that call with the Session Organ EKX. Yes, this might be seen as just a further expansion (EZkeys 2 is required to use Session Organ) but, given that the implementation must have required some adaptations to the EZKeys engine, I suspect this was a more significant undertaking than with a piano-focused EKX. Session Organ does, of course, have to find its way among a number of well-established virtual B3s available from other developers. So, is Toontrack's take on the Hammond B3 worth exploring?

Manual Control

While a piano and a Hammond organ are both keyboard-based instruments, they are clearly very different technologies. In the original Hammond, the sound was generated by spinning 'tonewheels' near a pickup, thus creating an electric current that was then amplified to drive a speaker. Based upon a well-preserved hardware example, Toontrack have combined detailed sampling of these individual tonewheels alongside some modelling as the basis for their virtual recreation.

The Hammond's twin 61-key 'manuals' (the upper and lower keyboard decks), alongside a set of 25 bass note pedals, are also reproduced here. The original hardware offers no velocity response (notes simply sustain until you release them) and volume/dynamics is controlled via an expression pedal. The same behaviour is obviously recreated here and, if you want to get the best out of Session Organ, a MIDI expression pedal (assigned to CC11 by default) would be an essential add-on to your MIDI keyboard controller.

As with all Toontrack's expansions, Session Organ ships with a broad selection of presets. I'll say more about these below but, within the Keys window, the Effects tab offers a preset-specific selection of additional macro-style controls to shape the sound. These can include tone, ambience, 'age' (for a more lo-fi sound), noise, options to mix different room mic levels, and a speed control for the Leslie speaker emulation. Some presets also include additional rotary options such as a Drive control.

However, the Keys window also includes an Organ tab and, if you pop this open, you get a view of the emulated drawbar controls, with sets of nine drawbars for each manual and a further two for the bass pedals. These essentially allow you to change the emphasis/balance between the root pitch and various octaves and/or harmonics of that pitch, giving you an authentic experience when shaping the tone of the resulting sound. This panel also provides access to various other sonic elements including the key clicks, percussive elements, vibrato and chorus, all of which are part of the original's character.

Adapted Engine

Given that the Hammond's drawbar system lets you individually customise the sounds from the two manuals and the bass pedals, you could see the original instrument as a three-part multitimbral sound source. Toontrack have customised elements of the EZkeys engine to accommodate this and, as a result, Session Organ supports multi-channel MIDI in/out. Those with multiple MIDI keyboards (and even a set of MIDI foot pedals) can therefore get the full Hammond performance experience. Alternatively, if just using a single »

The Organ tab within the Keys window provides full access to the emulated drawbars as well as other sonic elements.

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» MIDI controller keyboard, you can configure keyboard split points. And, as with the original hardware — which features sound preset keys at the left side of both manuals — Session Organ includes tonewheel preset switching via dedicated keyswitches that, by default, are mapped starting at C0.

To accommodate the multi-channel MIDI, Session Organ has suitable adaptations within the Grid Editor. You can still visualise and edit the MIDI notes for a performance within a single display, but located on the far-left side are options to solo/mute/isolate notes from the three individual layers if you just want to hear and/or edit a specific layer. This is neatly done and retains the familiar Grid Editor workflow despite the additional channels of data being handled.

The MIDI grooves included with the EKX support this three-channel format. If you export the MIDI from EZkeys to your DAW, it is automatically dropped as either three separate channels, or a single channel, depending upon your current configuration in the Settings panel as outlined above.

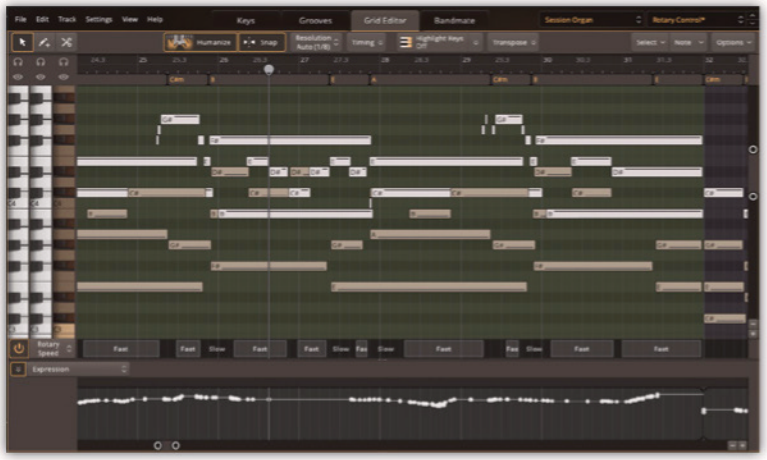
The other important adaptation to note is that the Grid Editor also includes dedicated automation lanes for the rotary speaker speed and the expression pedal, as well as other elements such as the drawbars or chorus/vibrato. The supplied Session Organ MIDI patterns fully utilise both of these. Session Organ does, however, work quite happily with MIDI patterns originally created for piano-based EZkeys content, but the velocity data will be ignored, and you do have to add any required expression.

Replace The Replacement

The original Hammond was primarily developed as an alternative to the pipe organ in a church context; small, less expensive and easier to maintain. However, it's still a big bulky beast that comes with an equally big bulky price tag and, as such, may be impractical or out of reach (or both) for the majority. Which is, of course, why virtual recreations are seen as convenient alternatives to hardware that was, originally, seen as a convenient alternative to even older hardware.

No, Toontrack are not the first virtual instrument developer to think this might be a good idea. And while EZkeys brings much more to the table than just the sounds, Session Organ does have to deliver on that front if it is to compete with the best of the existing bunch. Thankfully, it does... and this is ably demonstrated by the presets that are organised into Standard, Gospel, Jazz/Blues, Rock and Soul/R&B categories. A 'raw' DI preset is also included.

There are some truly wonderful sounds here that are full of character. The presets include obvious classic '60s pop references (Good Vibratos, Grey Onions or Pale White Shade, for example), but whether it's soft and warm, or bright and vibrant, Session Organ does it with ease. A personal highlight, however, was the classic rock tones; if you want to create your own Jon Lord moments, Session Organ has you covered. It sounds epic and the character of the original is authentically captured — including, in some configurations, the low-frequency hum that, in the absence of some notes within the same sub-200Hz range, is part of the full Hammond experience.



The Grid Editor supports the multi-channel format when editing Session Organ's supplied MIDI grooves.

Get Groovy Baby

The combination of the groove library and the Song Track makes EZkeys more than just a virtual instrument; it's a powerful virtual performer. The Session Organ EKX ships with a broad collection of MIDI grooves patterns and, while the instrument will happily play back grooves from other EZkeys libraries or MIDI expansion packs, those included with Session Organ are specifically designed to exploit the multiple MIDI channels and expression pedal options found here. They are a great way to explore just what the instrument can do if you are a bit of a Hammond B3 novice, and they cover all the musical styles found among the sound presets.

At the time of writing, Toontrack have also released a couple of add-on MIDI groove packs designed specifically for Session Organ: Gospel Organ and Classic Rock Organ. Having been supplied with the second of these two for the review, I have to say it is chock full of excellent stuff. If you want to add an instant touch of classic rock, psychedelic pop, blues, or progressive rock playing to your own projects, these MIDI groove packs are well worth their modest asking price. Here's hoping Toontrack have more style-based MIDI groove packs in the pipeline for Session Organ.

No Grinder, No Monkey, All Organ

The virtual Hammond B3 is not a new concept and many *SOS* readers might already have one or more of these within their virtual instrument collection. Personally, in sonic terms, I think Session Organ is absolutely up there with the very best of the existing alternatives. It offers a convincing recreation of the iconic sound and, because of the way Toontrack have adapted the EZkeys engine, it also provides access to all the key sound shaping options that made the original hardware such a versatile instrument. However, thanks to EZkeys' powerful virtual performer/bandmate and songwriting assistant functionality, Session Organ offers more than just being competitive in sonic terms.

The combination of great sounds, ease of use and powerful performance options make EZkeys 2 and the Session Organ expansion a heck of a combination. If you are already a dedicated EZkeys user, Session Organ is a truly excellent addition. And if you are just considering buying into the whole EZkeys ecosystem, Session Organ is a further great reason to take the plunge. Toontrack do not disappoint; the Session Organ EKX is beautifully done.

summary

Toontrack's Session Organ EKZ is beautifully done, capturing the sounds and features that made the original Hammond B3 such an iconic instrument. A great addition for EZkeys owners.

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Plugin Boutique Scaler EQ

EQ Plug-in

This innovative plug-in applies a little music theory to the humble equaliser.

JOHN WALDEN

Whether it's authentic recreations of hardware classics, multiband dynamic or AI-powered, our EQ options have come a long way since the humble treble and bass controls of days gone by. But Plugin Boutique's Scaler EQ adds a new twist to the EQ knob: this is an EQ that, thanks to its awareness of the key/scale of the music you're processing, can apply adjustments focused on just the notes/frequencies in (or out of, or both) your project's key. It's the kind of idea that, once you've heard about it, makes you wonder why it isn't already a thing!

Lightbulb On

In brief, Scaler EQ is a six-band EQ plug-in that supports a full range of traditional EQ options: peak (parametric), low/high cut and low/high shelf, with both fixed and dynamic operation, stereo, Mid or Sides options, and external side-chain support. There are also Magic Shelf (both high and low) options that offer additional tweaks for improved results with stereo signals. The plug-in provides a comprehensive suite of controls in each case, so you could easily use Scaler EQ for any of your routine EQ duties. It also includes spatial width options and input/output controls with their own limiters for managing your levels.

But the really novel option is Harmonic Peak mode. The control set is identical to that of the peak mode filters but, as can be seen in the screenshot, a Harmonic Peak band consists of multiple individual filters positioned around the centre frequency of the band. The position of the individual filters is determined by the frequency of the notes that are in (or out) of the currently selected key/scale combination. The Detect Key option (located in the top toolbar) will automatically detect your project's key, or you can specify it manually. The Scale Switcher buttons allow you to store up to three key/scale combinations, and you can use automation within your DAW to switch between these. As you do so, any Harmonic Peak bands will follow, adjusting the frequencies of those narrow individual filters to match the notes within the scale.

While you can change this behaviour, by default, when you boost with a Harmonic Peak band, the individual filters focus on notes within the chosen scale — and when you cut, the filters focus on notes outside the scale, so as to reduce any harmonic content that's out of key. In addition, as for band four in the screenshot, there is a Harmonic Peak +/- mode — this combines these two behaviours in a single band. And, yes, the Harmonic Peak modes can also be dynamic.

Beyond The Concept

Even as a flexible conventional EQ plug-in, Scaler EQ is capable and modestly priced, but it's definitely the scale-aware EQ'ing that makes it stand out from the crowd. There's certainly an appealing logic to the underlying idea but I think the practical potential of this plug-in is just as interesting as the concept. And having experimented with the plug-in on a number of projects during the review period, I'm still not sure I've fully grasped its full potential, partly because it is a novel approach but also because it opens up some equally novel possibilities.

You can, of course, use it to sweeten up a vocal, individual instrument or a bus. In that context I found that a Harmonic Peak band could subtly tame those 'out of key' frequencies to deliver a little extra clarity, though you do have to use it with care — if applying too much gain, I could hear some artefacts creeping in. Scaler EQ also has strong potential as a sound design tool. For instance, if applied to a percussive sound that has little obvious harmonic relationship to the project's key/scale, a Harmonic Peak band can emphasise the 'in key' frequencies, so that the percussive sound becomes 'tuned'. I'm sure that, given more time, I could unlock more of its potential, so I'd absolutely encourage you to try the free trial version for a spin.

summary

Scaler EQ is a well-specified modern EQ plug-in but with an additional — and very novel — key/scale awareness. Well worth exploring.

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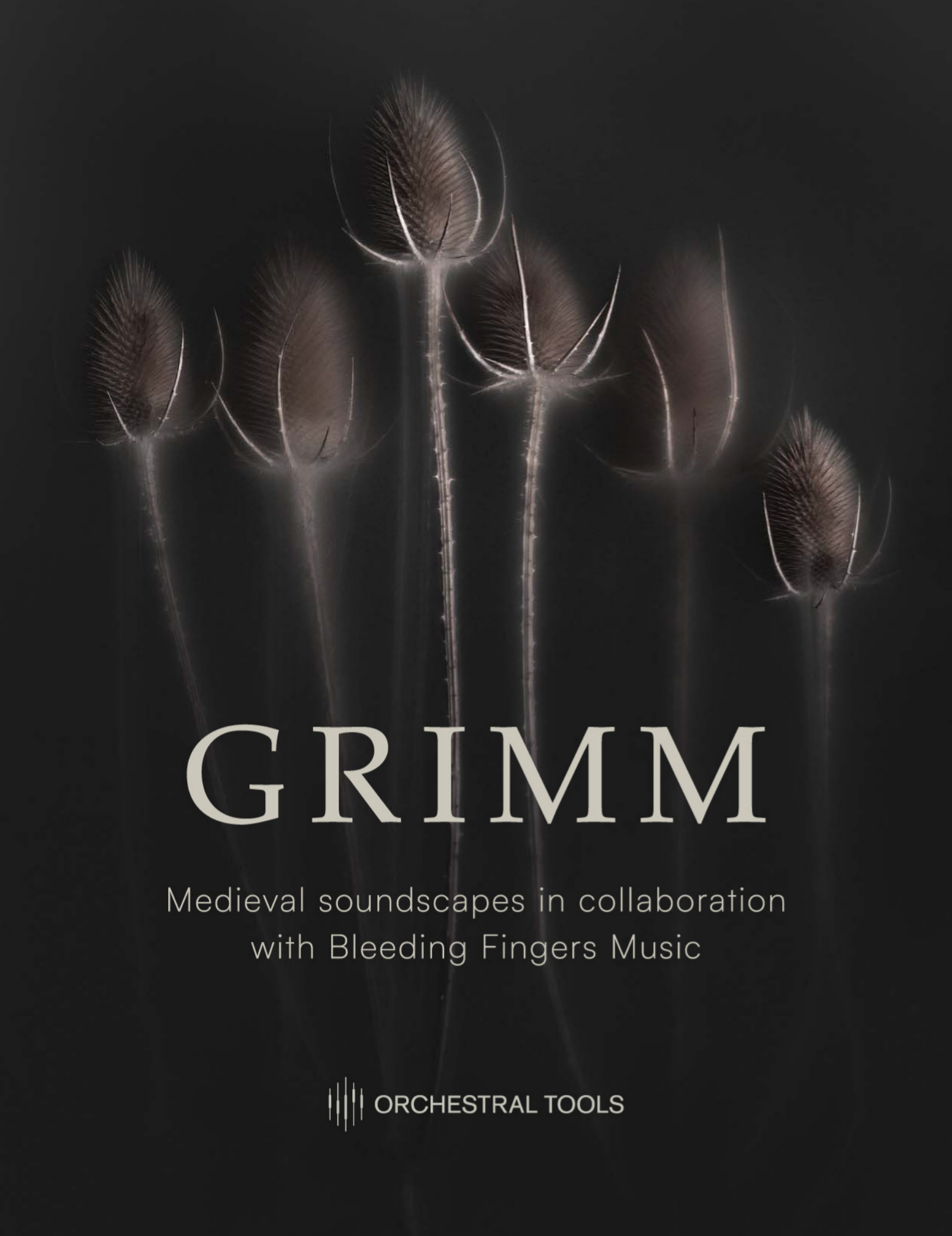
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Warm Audio WA-19

Dynamic Microphone

SAM INGLIS

For their latest vintage recreation, Warm Audio have resurrected a dynamic mic revered by Beatles fans and audio engineers alike.

Recreating iconic studio products from the past is now big business, but it's a business that tends to focus on the same few products. If you want a microphone with the number 47, 67 or 251 in its name, you're spoilt for choice. But if you want a microphone closely associated with the Beatles and responsible for most of Ringo's early drum tracks, you're at the mercy of eBay and Reverb. That is, until now.

Warm Audio's catalogue already includes homages to the U47, U67, ELA M 251 and all the other familiar classics, and they are now expanding into areas where there's less competition. Last year, for example, they introduced preamps based on the Ampex 351 — and in 2024, they've tackled that legendary Ringo overhead mic. The WA-19 recreates the AKG D19, a moving-coil dynamic model that was a mainstay of stage and studio throughout the 1960s.

Back in May 2021, I attempted to piece together the history of the D19 for an *SOS* feature. I won't reproduce that entire story here, but if you're interested, it's freely available to read at www.soundonsound.com/reviews/golden-gear-akg-d19. For the purposes of this review, a brief summary will have to do!

Introduced about 1957 and discontinued towards the end of the following decade, the cardioid D19 was a highly popular mic which went through several iterations, and was rebadged by numerous other manufacturers and distributors. It was notable for its compact size, relatively hi-fi frequency response, and for AKG's implementation of something analogous to Electro-Voice's Variable-D technology for reducing proximity effect. Low-frequency control was further aided by an unusual mechanical high-pass filter, switched by rotating a ring at the base of the capsule. The D19 is sought after today partly because of its association with the Beatles, but also because it has a distinctive, airy sound that can be somewhat like that of capacitor mics, especially on acoustic guitar.

N-N-Nineteen

Warm Audio don't offer the same confusing array of variants with the WA-19, thankfully, but it is available in a choice of nickel and black finishes. The nickel version was supplied for review, and its resemblance to the original is uncanny. From the texture of the paint to the pattern of the grille and the action of the rotating filter switch, Warm Audio have got the look and feel absolutely spot on. And the similarities go much deeper: developed specially for this remake, the capsule assembly is a dead ringer for the original, to the point where Warm Audio say that it fits in the D19 shell and will be available as a spare part for reanimating the many dead AKG mics out there. This is great news.

Look closely, though, and there are a few differences in the detail. The screws that hold things together are larger than on the D19, and Warm Audio have added a discreet badge to the front of the grille. The filter ring works 'backwards' compared with the original, and they've replaced the M(usic) and S(peech) legends with the standard symbols for flat and high-pass filtered responses, which seems sensible. Only one change strikes me as potentially significant, though. The capsule vents into a narrow brass tube running the length of the mic, as it should. On the original, this tube is housed in »

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» a channel within a second brass tube, which is omitted from the WA-19. Warm Audio told me that this second tube existed purely to strengthen the shell of the original, and that it isn't needed on the WA-19 because of improvements to the design of the body.

The WA-19 is a transformerless mic with a 200Ω output impedance, which is both significant and faithful to the original; the D19 was available in 50Ω, 200Ω and 50kΩ variants, and the 200Ω model was indeed transformerless, with the capsule connected directly to the output. The D19 was sold with multiple different output connectors, but I've never seen one with the now-standard three-pin XLR; this, of course, was the only sensible choice for the WA-19, and means that it's perhaps half an inch longer than the classic D19C. Copying the D19's charming wooden box would no doubt have added quite a bit to the cost, so the WA-19 is supplied in a pouch, with just a mic clip for company.

Up Close & Personal

Back in 2021, when I wrote the article referenced earlier, my studio was home to six D19s, plus a couple of related models. Shortly after that feature appeared, however, I got talking to the mic technician at a well-known studio, who told me that the D19s they'd had since new were now failing and unrepairable. As a result, my collection now has a new home, and I've only managed to pick up one working D19 since. No two of the D19s I owned sounded the same, but there is a kind of sonic signature that is common to all the working D19s I've ever used. It's a mic that really loves being close up to the source; there's a lot less proximity effect than you hear in most cardioid mics, and it's not blessed with tons of low end to start with, so you can jam it right up to an acoustic guitar or other instrument without things getting at all muddy or boomy. With a presence lift that's higher up the frequency spectrum than those of most dynamic mics, it gives the impression of detail whilst suppressing all those annoying pick noises and other transients that capacitor mics deliver when used close up. It's simultaneously bright, yet quite soft-sounding.

Talking to the team at Warm Audio, it seems that their experiences with the vintage mics were similar to mine. So, rather than use any single surviving D19 as a model, they have tried to extrapolate backwards from multiple examples to recreate what they think a D19 might have sounded like when new. In terms of frequency response, they say this means the WA-19 may have slightly more going on at either end of the spectrum than a typical aged D19.

Spot the difference: the WA-19 (right) with an original AKG D19C.

In their publicity materials, Warm Audio say the mechanical filter in the D19 design doesn't introduce phase-shift in the way electrical filters do. My physics isn't up to evaluating that claim, but I had the opportunity to compare my D19 and the review WA-19 in a small test chamber; and, curiously, there was a noticeable difference in LF phase and frequency response between the two with the filter switched *out*. Whether that missing brass tube has a role to play here, or ageing has just taken its toll on mine, I can't say.

In any case, I wouldn't expect the WA-19 to measure or sound identical to my D19, but the important thing is that it approximates the general character very effectively. You can get intimate with it without low-mid bloat becoming a problem, and you can strum fiercely without things getting harsh. It puts across an impression of air and detail in the treble range without being the least bit harsh, or emphasising unwanted micro-transients. The review mic perhaps had a slightly harder-sounding midrange than my D19, but it's right in the ballpark.

Fresh Vintage

Manufacturers who specialise in reproducing classic gear attract their share of critics bemoaning a lack of innovation, or questioning the morality of copying other people's designs. In some cases I share these concerns, but I'm delighted with the direction Warm Audio have taken here. The last D19 rolled off the production lines more than 50 years ago. The factory, tooling, staff and designers that made that mic are all long gone, and there's little chance of AKG reissuing it. The only people who will be hurt by the WA-19 are the profiteers of eBay and Reverb, and few will shed a tear for them.

Warm Audio have clearly put a huge amount of work into the WA-19, and the fact that they've brought it to market at such a reasonable price is highly impressive. And, strange though it sounds, it really does tread new ground, at least as far as recreating vintage gear goes. There have been innumerable copies of classic capacitor and ribbon mics, but to my knowledge, this is the first serious attempt to replicate a vintage moving-coil model. Now that Warm Audio seem to have the expertise to do that, what will they bring us next?

summary

Warm Audio have recreated the classic AKG D19 with impressive attention to detail!

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iZotope RX11 Advanced

Spectral Editing & Audio Repair Software

We check out the latest iteration of what arguably remains the most comprehensive audio repair software out there.

RX11 brings a number of improvements to both the Dialogue Isolate and Dialogue Contour modules for processing voice-based audio.

iZotope RX11 Advanced
$1199

PROS

- RX11 expands upon an already comprehensive audio repair toolkit.
- Some very useful improvements for those working with dialogue, stem rebalancing or loudness management.
- M-S mode a simple but potentially powerful addition.

CONS

- Not all existing users will find the new/improved features essential.
- Price of the Advanced edition might put it beyond many.

SUMMARY

RX11 brings more useful tools to iZotope's already impressive audio repair software. It's a specialised application with a price to match, but the Standard version will hit the sweet spot for all but the most demanding users.

JOHN WALDEN

Thankfully, spectral editing is not what it used to be! Yes, you can still do those forensic-level manual edits within the spectral display where that's what's required and if you have the skills, but with easy-to-use processing tools and some clever machine-learning algorithms under the hood, products such as iZotope's RX now allow anyone to access the power of this kind of editing.

RX remains one of the leading products in this field and, some 18 months after Hugh Robjohns assessed RX10 (*SOS* January 2023: https://sosm.ag/izotope-rx10), iZotope have now delivered RX11. As before, RX11 is available in three versions, Elements, Standard and Advanced, that at heart are the same app but with more or fewer features for different prices. For full details on those differences, you can check out the comparison chart on iZotope's website (www.izotope.com/en/products/rx/compare), but the main focus of this review will be the 'full fat' Advanced version.

iZotope have presented recent iterations of RX as a 'toolkit for audio repair', and that toolkit comprises a collection of processing modules, each designed to streamline a specific editing task, such as noise reduction, de-clicking, reverb removal, reducing guitar finger squeaks, matching EQ from a source to a target, or stem unmixing/rebalancing. The quality of result can, of course, depend on the nature of the audio being processed, but more often than not RX is capable of dramatically improving poor audio, and even nudging 'unusable' audio over the threshold into usable territory. As well as building on the functions of RX10, RX11 integrates further advances in machine-learning technology and adds some headline 'new and improved' features.

Voice Control

RX11 brings a number of new and improved tools for processing audio containing »

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the human voice, and perhaps the most significant is a reworking of the Dialogue Isolate module. With machine learning underlying the processing, this now combines both de-noise and de-reverb options into a single, simple-to-use module that can be applied in real time (the modest amount of latency is perfectly acceptable for post-production editing). What's more, this facility is, for the first time, available in RX Standard (using the real-time algorithm), while RX Advanced's version adds a further high-quality (albeit offline rather than real-time) option as well customisable multiband processing.

In my tests, Dialogue Isolate did a remarkable job of clearing away the sonic clutter in some old-school dialogue recordings and, importantly, it did so without compromising the quality of the spoken voice. There are obvious applications in journalism and broadcasting, but I can also easily imagine podcasters using Dialogue Isolate as part of their post-production workflow for that final bit of clean-up. That said, if this is all you want a tool to do, there's plenty of competition, including iZotope's own VEA and Accentize's dxRevive (with the latter being particularly impressive).

Available in RX Advanced's standalone Editor (not as a separate plug-in), the Dialogue Contour module has also been improved. As well as a reorganisation of the various slider controls, this includes a pitch curve that's superimposed on the waveform, giving you a very useful visual representation of the rise and fall of 'expression' within the dialogue, and as you add/edit pitch-curve nodes in the spectral display, a second curve shows how your changes affect the pitch curve. The control set also now includes a Variation slider that allows you to either expand (more expressive) or contract (less expressive) the overall pitch variation in this curve. This can be really helpful if you need to finesse the expressivity in dialogue or, for example, smoothly drop in word edits from multiple takes or different points in a recording.

The Repair Assistant also benefits from improvements. This is available in all editions of RX but, of course, the signal chains it creates for you from the initial 'learn' process is restricted to the module components of the version you own. You can choose between processing voice (dialogue or sung) or instrument sources and, as shown in the screenshot, the module/plug-in then provides a compact set of initial controls across four panels.

The Repair Assistant has also been improved in RX11.

If required, you can open the full module chain for more detailed control. While this approach is very flexible, it does seem fairly CPU intensive. So if working on a modestly specified machine, you'll probably want to render your Repair Assistant edits before moving on to the next editing task.

Unmixing Magic

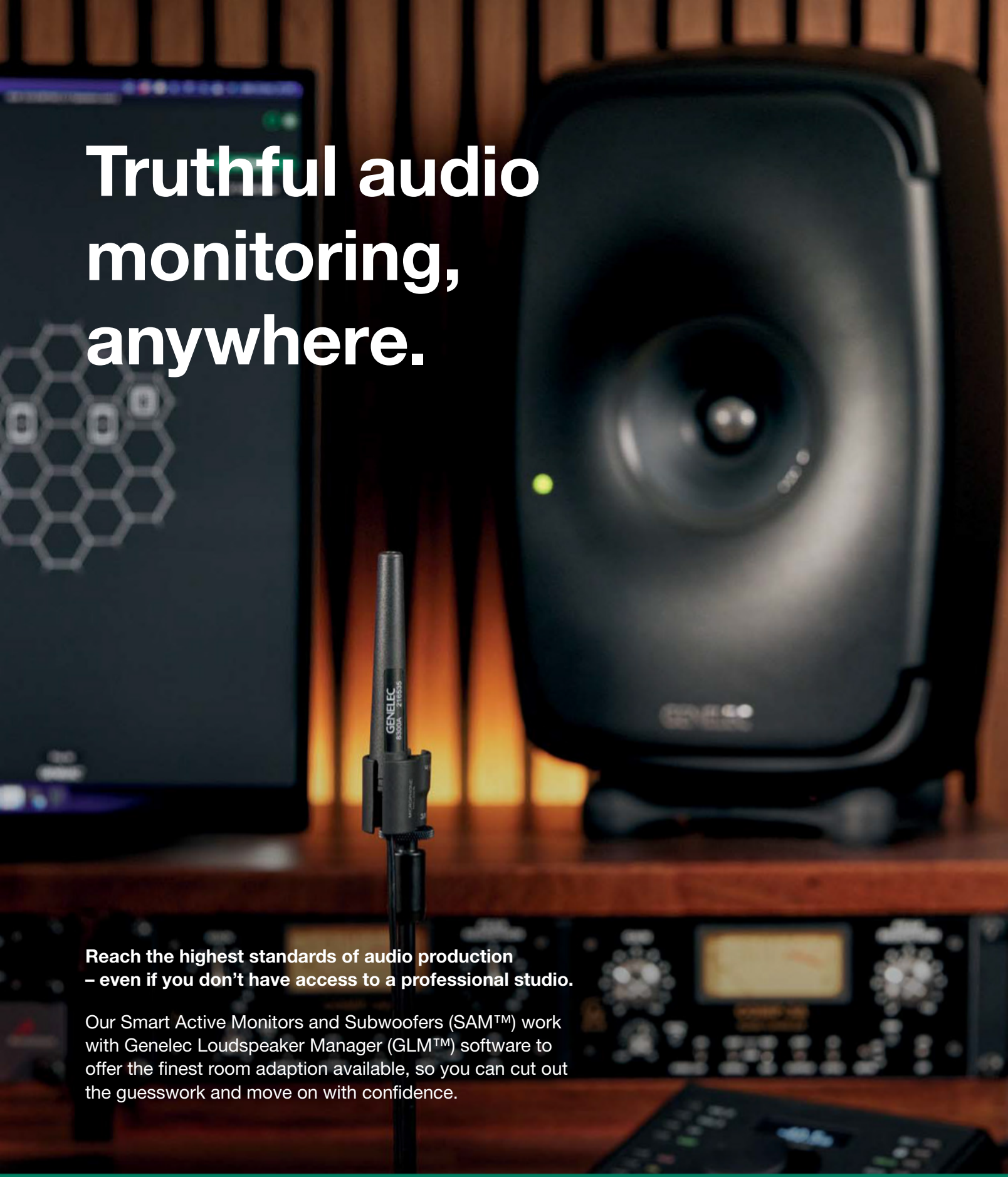
Music Rebalance, RX's stem separation tool, also benefits from improved machine learning and there are some useful refinements to the GUI and control set too. RX10's global Separation slider has been replaced by individual Sensitivity sliders for each of the four stems (Vocal, Bass, Drums, and Other), so you can now fine-tune the relative emphasis the separation algorithms place on these four sounds. In practice, this translates to the user enjoying an extra degree of control that means you can focus on a specific target should you wish.

Creating instrumental mixes or backing tracks (in other words, removing vocals) is almost always achievable to a remarkably high standard. More problematic – even if it's still something approaching sorcery! – is the opposite: isolating a vocal to create an a cappella. Busy mixing, lots of reverb/delay on the vocal, or the presence of backing/harmony vocals, make generating a clean result more challenging. However, the Sensitivity slider does give you some room to experiment on that front and, given the additional voice repair options RX offers, there are options for some further cleaning (de-reverb can be really helpful, for example). I should mention that isolating harmony and backing vocals remains a challenge: whoever cracks that particular bit of machine learning/AI first will undoubtedly jump ahead of the competition. And speaking of competition, while RX is impressive it's not the only software tool capable of this sort of unmixing magic; I'd say that the likes of Steinberg's SpectraLayers and Hit'n'Mix's RipX are capable of comparable results.

Central Perk

There's one further new feature in RX11 that might also have some potential when it comes to vocal isolation, among other things: the addition in both the Standard and Advanced versions of a Mid-Sides editing and processing mode. For stereo files, in the View menu you can now flip the channel mode between the conventional Left-Right stereo and Mid/Sides. Once you've done that, all RX's modules can process the Mid and Sides components of a stereo signal separately. This could open up all sorts of

The new Sensitivity controls allow you to encourage Music Rebalance's sophisticated stem unmixing algorithms to focus on a specific component within the separation process.



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The new Loudness Optimize module lets you easily apply upwards compression to the quieter sections within your audio project to help avoid any undesirable adjustments being made by subsequent loudness normalisation.

» applications, but when it comes to isolating a vocal you'll often find that the lead vocal sits mainly in the Mid channel, while the Sides contains a little more of the ambience, effects and backing vocals. It might be a marginal gain but if you're looking to extract a vocal part for a remix, this would be well worth exploring alongside Music Rebalance.

There might also be potential here for getting creative with the bass or drum stems created by Music Rebalance, whether it's to refocus bass frequencies in the middle of the stereo image, or to add a little extra stereo sparkle to a drum stem. The same Mid versus Sides content variation might also apply with dialogue in stereo field recordings (particular noise elements might well be more prominent in the Sides). Anyway, it's a great addition and will, I suspect, get lots of RX users thinking of new ways it might help with routine repair and editing tasks using the existing suite of modules.

ARA: An Open DAW Policy?

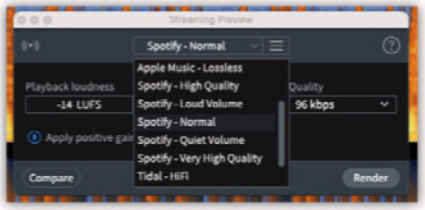
As with RX10, there's an ARA plug-in version that, currently, works only in Logic Pro (in Rosetta mode), but iZotope tell me support for other DAWs is in development, with Pro Tools and Studio One support due in a free update during Summer 2024 and support for other DAWs to follow. The Logic version certainly worked well on my test system, and I enjoyed a tighter sense of integration than when using RX as an external application — it's a work in development, but I'll be interested to see how far iZotope can take this.

Streaming & Loudness

RX10 Standard and Advanced included a Loudness Control module, while Advanced also offered the Leveler. These modules remain and each bring something to the table in terms of loudness management but in RX11 Standard and Advanced iZotope have upped the game considerably with Loudness Optimize and Streaming Preview.

Loudness Optimise provides a visual and numerical summary of a file's loudness. If required, it can then apply upward compression to the quieter sections to increasing the average loudness while leaving the louder sections and peaks unaltered. The display includes a representation of the 'weighted gate' that's used in the BS.1770 Integrated Loudness calculations. Any audio that falls beneath this gate is automatically excluded from the integrated loudness calculation (which is why, for some audio files, the '% measured' value shown top-left will be less than 100%). In calculating average loudness, the gate means sections of silence or low-level material, not least within TV/film broadcast content, do not cause 'misleading' loudness measurements, ensuring a more consistent result for the listener.

For music tracks aimed at streaming platforms, though, where significant passages in your track fall below this gate, this can have a marked impact on how the loudness normalisation of each streaming platform is applied to your track. Loudness



The new Streaming Preview module lets you hear how your music will sound once subjected to loudness normalisation on any of the major streaming platforms.

Optimise therefore provides a great way to spot this potential problem. Then, through a combination of its various controls, you can apply some upward compression and then preview (and render) a modified version of the track that's less likely to fall foul of the loudness normalisation applied during streaming. It's really easy to see and hear what's going on.

Streaming Preview complements this very nicely. It allows you to audition your mixed and mastered track with the loudness normalisation processing that would be applied to it by specific streaming platforms. You can also render a version of it that includes that processing. Currently available presets include variants of Amazon Music, Apple Music, Spotify, Tidal and YouTube, but you can configure a specific loudness, codec and audio quality combination manually if you need to. It's a super-useful tool that allows you to determine whether, in terms of its loudness, your track is ready for upload or still requires some finessing.

Take It Up To 11?

So, is RX11 for you? Well, as Hugh Robjohns suggested of RX10, it's fair to describe this as a case of evolution rather than revolution, but it's worth reminding ourselves just how remarkable RX11 has become. Yes, it's a specialist tool that not every audio editor or music creator will feel compelled to use, and yes, there are some apps that might outperform it on some individual tasks. But if your work routinely involves rescuing audio that is sub-optimal, RX is one of the very few comprehensive options that you might turn to in order to save the day. In RX11, iZotope have pushed the limits of what's possible that bit further and made doing it that bit easier. For many pro audio editors regularly trying to do the impossible to a tight deadline, that may be all that matters.

$ RX11 Advanced $1199. RX Standard $389. RX Elements $99. (Discounted to $799, $299 and $49, respectively, when going to press)
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SAM INGLIS

I'm not sure how Antelope Audio came to choose the name Zen for their range of desktop audio interfaces, but there's a sense in which it's quite apt. For most of us, I think, the ideal audio interface is one that we're not really conscious of. The less we have to think about it, the better, and a perfect audio interface should offer complete sonic transparency and intuitive ease of use.

Antelope have spent many years pursuing the first of these goals, and their interfaces have consistently claimed class-leading specifications. Much thought has also gone into how the user interacts with the interface, through features such as hands-on controls and, on some models, touchscreens. Meanwhile, in a less obviously Zen move, they've pioneered the inclusion of FPGA- and DSP-based digital effects that can be applied in the recording path, offering low-latency EQ, dynamics, preamp emulation and much more.

Antelope Audio
Zen Quadro Synergy Core
$599

PROS
- Good selection of I/O including secondary USB port for phones and tablets.
- Impressive audio specifications, especially for a bus-powered interface.
- Built-in Synergy Core signal processing with a generous range of AFX modules included.
- Switchable DC coupling.
- Excellent value for money.

CONS
- AFX are always in the record path, and can't be hosted as DAW plug-ins.
- Only basic monitor control features.

SUMMARY
Antelope Audio rewrite the value equation with an interface that offers class-leading audio specs at a highly competitive price.

Q To Quadro

The latest interface to bear the Zen name is the Zen Quadro Synergy Core. Viewed from the top, it's pretty much identical to the earlier Zen Q Synergy Core USB interface reviewed in May 2022 (www.soundonsound.com/reviews/antelope-zen-q-synergy-core-usb), with the same large and slightly clunky rotary encoder, small push-buttons and colour display (which is not touch-sensitive). But whereas the Zen Q has an angled body, the Zen Quadro's is flat, and although both offer four analogue inputs, the newer model has four mic preamps to the Zen Q's two. The front-panel combi inputs can still operate in high-impedance mode for guitar recording, of course.

On the output front, the Zen Quadro follows the Zen Q in offering two pairs of line-level quarter-inch jacks, plus two headphone sockets. It also retains the Zen Q's complement of digital I/O, meaning that there are coaxial S/PDIF in and out sockets plus a single optical input for ADAT Lightpipe. And like the Zen Q, the Zen Quadro is bus-powered over USB, with no mains PSU socket and no Thunderbolt option, but there's a significant development here in the shape of a second Type-C USB port. This allows other USB audio-capable devices such as phones or even a second computer to be integrated. The Zen Quadro is a class-compliant device, which uses the familiar Thesycon driver on Windows and the built-in Core Audio driver on macOS.

Light Launch

A key focus in Antelope's drive to improve ease of use over the years has been the Antelope Launcher software utility. This doesn't catapult kudu or hurl hartebeest, but acts as a central hub for everything related to Antelope hardware

and software, handling installation, updates, authorisation and more. Once this is installed, getting started should be straightforward, but in my system it wouldn't acknowledge the review Zen Quadro until I'd visited the System tab and rebooted the Manager Server.

Though it's a separate program, the Zen Quadro's software control panel can be opened only from the Launcher. Assuming you want the control panel open while you work, this has to be done each time you boot your computer, though it's possible to save settings within the hardware and even use it standalone if you wish.

The control panel presents three separate mixer panes, respectively feeding the main monitor outputs, mirrored at the first headphone output; the second headphone output; and the second pair of line outputs. There are also routing panes that allows inputs to be assigned directly to outputs. This is useful, because the primary USB port presents 16 ins and outs to your DAW; with up to 16 physical inputs also available, this means there are twice as many potential sources as there are channels in the mixers.

Each mixer pane offers the same selection of 16 mono or eight stereo channels, which are initially drawn from the four preamps, the first pair of playback channels from the primary USB device, stereo playback from a secondary USB device, and the eight ADAT input channels. Sources such as the S/PDIF input are absent by default, and if you want to feed them into the mixer, you'll need to sacrifice other sources in their place. As well as S/PDIF and additional USB playback channels, other options include 'emulated mic' channels for use with Antelope's Edge mic modelling system, and a choice of two test oscillators. It's also possible to set USB record inputs to pick up 'loopback' signals from the mixer.

The AFX Team

The first six sources in the mixer have insert points for the onboard AFX: signal processing presented in a plug-in-like interface, but running on the internal Synergy Core platform. The Zen Quadro comes pre-equipped with a healthy selection of 37 AFX modules, which encompass guitar amp and cabinet emulation, modulation effects, vintage-style EQ and compression, preamp emulation, a tuner, and more. Many additional AFX are available as add-ons. These channels each have eight AFX 'slots', which are individually bypassable, and AFX can be reordered by dragging and dropping. There's no preset structure for individual AFX, but it's possible to save entire chains as presets, including the settings of the individual modules. A number of preset chains are supplied.

Conceptually, the Synergy Core system has something in common with Universal Audio's UAD2 Powered Plug-ins platform, but there are some differences. Although there is a global Auraverb reverb, AFX doesn't otherwise offer time-based effects such as delays and reverbs; and whereas UA's Console makes it possible to use UAD2 processing only in the monitor path on a per-channel basis, AFX are designed to be printed with the source. (It is, however, possible to route each preamp simultaneously to a second USB record destination if you want to record it dry.) The Zen Quadro doesn't support Antelope's AFX2DAW plug-in, which would allow AFX to be used at mixdown »

» within your DAW's insert points in the same way UAD plug-ins can be, but you can route USB playback channels from the DAW through AFX and re-record them or have them running live at mixdown. The AFX devices themselves are generally of high quality, and the collection that comes free with the Zen Quadro is more generous than the Realtime Analog Classics bundle you get with UA's desktop Apollo interfaces.

There's no meter that reports what proportion of the Synergy Core processing has been used up, but as I understand it, this isn't as simple an equation as it is with a purely DSP-based platform like the UAD2 system. At any rate, I never managed to reach the limit in normal use, even with multiple AFX instantiated on all six channels. In general, the system is well thought-out, easy to use and genuinely useful; my only tiny gripe is that if you have the control panel window in Compact mode, it's not possible to tell at a glance whether a channel has AFX loaded or not.

Core Strengths

The Zen Quadro boasts remarkable specifications, especially considering that it's a bus-powered USB device. The mic preamps offer a 75dB gain range, in 1dB steps, with THD quoted as -116dB — equivalent to 0.00016 percent. A-weighted dynamic range is given as 122dB on the analogue inputs and a colossal 130dB on the monitor outputs. These are the sort of specs you'd expect to see on a mastering-grade converter costing thousands, not a squat black desktop box retailing at only a few hundred. It seems unlikely that most users will ever need quite that much dynamic range, but it certainly fulfils the remit of allowing you to forget about what the converters and preamps might be doing to your sound.

The control panel and routing possibilities strike a good balance between flexibility and ease of use, and although I can imagine circumstances where it might be restrictive for example to have the main outputs always mirrored on the first headphone output, 99 percent of the time it'll be exactly what you need. Four analogue inputs with mic preamps is enough to cater for a broad range of recording scenarios, and the ADAT input makes it easy to expand when you want to track drums or similar. Though it's stereo-only, the second USB input works very well when you need to connect a phone or other consumer source; just be aware that if you have an older iPhone with the Lightning connector, you'll need to use a proper OTG adaptor such as the Apple Camera Connection Kit — a generic USB-to-Lightning cable won't work. One area where some rival interfaces score over the Zen Quadro, though, is monitor control: there's independent volume and mute for the monitor outputs and the two headphone outputs, but no dedicated mono button or one-click speaker switching.

The most impressive thing about the Zen Quadro Synergy Core is the value for money it offers, as a glance at the competition confirms. Arturia's AudioFuse Studio offers more versatile I/O options and also has impressive specs, but is significantly more expensive, as is Avid's MBox Studio; and although MOTU's M6 is more affordable, it lacks the Zen Quadro's digital I/O — and none of these has built-in signal processing on the same level. If you need that, Universal Audio's Apollo x4 has a similar basic configuration, with four mic preamps and two headphone outputs. It offers an additional pair of line outs, ADAT out as well as in, a nicer action on the encoder, the benefits of Thunderbolt connection and an effects platform that is arguably more versatile than the Synergy Core — but it's more than twice the price of the Zen Quadro.

In short, although some other desktop interfaces may have an edge in terms of ease of use, I/O count or versatility, you pay a lot more for that edge. None of them betters the Zen Quadro's audio specs, and none competes on price. The Zen Quadro Synergy Core offers a combination of features and sound quality that is unmatched for the money, and I'm sure it will be a big success.

The Zen Quadro's control panel, with the AFX editor for channel 1 open in the pop-up.

Down To DC

As well as handling mixing and routing, the Zen Quadro control panel also offers numerous configuration settings. Some are obvious, like switching the preamps between line, mic and high-impedance mode, setting gain and turning phantom power off and on, but there are some more surprising options lurking in the pop-up that appears when you hit the cogwheel icon. These include level and frequency settings for the test oscillators, alignment levels for the analogue outputs and, most notably, the ability to switch DC coupling on and off independently for the inputs and outputs. The ability to accept and output DC voltages is useful for integration with modular systems, and is not uncommon these days, but I've never before come across an interface that gives you the option.

Incidentally, although mixer settings and AFX can only be changed from the control panel software, many other parameters including all of the above-mentioned ones are accessible from the Zen Quadro itself. Pressing and holding the Gain or the HP/Mon button takes you into Control and System menus respectively, which between them span everything from choosing a clock source to setting up input-to-output routing.

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Drawmer Electronics 1971

Dual-channel Parametric Equaliser

With a compressor-saturator stage for each of its four bands, could Drawmer's new EQ become your latest Crush?

MATT HOUGHTON

Last month I took a look at Drawmer's 1977 channel strip, and enjoyed it immensely: a supremely versatile design, it incorporates a clean mic preamp, a three band EQ section, a FET compressor and variable saturation, so it offers pretty much everything one might usually look for in a channel strip. But sometimes you need a little more control than you can realistically pack into a channel strip, and an extra channel can come in handy too. Which is why Drawmer's 1970s range also incorporates several rackmount processors that share the 1977's sound but can be used to do that bit more. The latest to join the line-up, which already featured a variety of preamps, EQs and FET compressors, as well as a width and saturation processor, is the new 1971 EQ, which is reviewed here.

Features

Designed and handmade by Drawmer in the United Kingdom, the 1971 is not your average EQ. I mean, it *can* be used that way, but this really is a unique design. At heart, it's a dual-channel four-band parametric EQ that Drawmer describe as taking its "inspiration from 1970s-era analogue gear". Each channel has an identical set of controls that you can use for two mono signals or one stereo one, and you can even patch one channel into the other if you need greater control over a mono signal. In addition to the four parametric bands, there are one high- and one low-pass filter, both with a generously wide frequency range (from 10-240 Hz and 4-31 kHz, respectively) and, like the other bands, these have their own engage/bypass switch (a hardwired bypass), making A/B comparison easy. Helpfully, there's also a ±15dB gain stage on the left, to allow you to adjust the signal level before it hits the equalisation, and there's a similar control on the right to adjust the output level, and these each have a horizontal, five-LED, post-gain level meter.

But the headline feature that really makes this EQ stand out from the crowd (I don't remember seeing it in any other equaliser, even Drawmer's own 1974 stereo parametric EQ), is the Crush facility. This is a combination of a fixed time-constant compressor and saturator that can be switched in/out individually for each of the four parametric bands on each channel. This processor has an auto make-up gain, and it's responsive to the amount of gain in each band, becoming somewhat interactive when Crush is engaged on multiple bands on the same channel. Drawmer tell us that "it'll fatten your bass, enhance the presence of your mids, and bring out the shimmer in the top end". The few weeks I spent putting this EQ through its paces confirmed that their description is accurate! In practice, I often found myself wanting to use this feature more subtly and selectively, usually choosing to enhance just one band, though I'm sure many others will enjoy squeezing every last drop of attitude out of it!

The outer two EQ bands (ignoring the high-/low-pass filters) are shelving types with Q controls similar to those on the 1977 channel strip. As with all knobs on this device, those here are detented pots that make it easy to match and recall settings, while offering more options than on most switched selectors. The channels seem well matched too. The low band's frequency control ranges from 35-700 Hz and, as with all other bands, there's ±12dB of gain available. Unlike the two bands in the middle, there's no Octave knob to set the

Drawmer Electronics 1971

$1799

PROS

- A very capable equaliser, with high- and low-pass filters.
- The Crush feature is special!
- Clear labelling.

CONS

- Nada!

SUMMARY

Don't let the unassuming appearance deceive you: with it's Crush facility, the 1971 is a very special EQ.

bandwidth. Instead the low band has two buttons, combinations of which (indicated using two yellow LEDs) set the steepness of the slope: 6, 9 and 12 dB/octave options are joined by a fourth setting, which turns the shelving band into a peak filter, which can be incredibly useful when used alongside the high-pass filter. The upper band, meanwhile, has a single button to toggle between 6 and 12 dB/octave; the latter, as on the 1977, not just being steeper, but also giving you a handy little dip before a boost.

The two bands in between, labelled Low Mid and High Mid, are the same except that they have an Octave knob to set the bandwidth. As almost always on Drawmer devices, there's a generous range, with the Q going from a very narrow 3.3 out to a much gentler 0.33, which means it's pretty useful for taming ugly resonances as well as more general tone-shaping duties. There's also a helpful overload LED next to the high shelf band (there's one for each channel) which tells you if you are clipping and need to dial down the EQ or input gain.

Generally, across the front panel, the labelling is in Drawmer's usual no-nonsense, easy-to-read white on black. Around the back, you'll find two pairs of XLR connectors for the balanced line-level I/O, and an IEC mains inlet for the internal, low-hum linear power supply (which can be configured by the user for 110 or 230 Volt mains). You'll find more specifications on Drawmer's website and in the manual, but suffice it to say that despite the 1970s inspiration, modern components and methods are used and the result is a low-noise, low-crosstalk design with a near-as-damn-it flat frequency response when you're not boosting/cutting.

Crush Hour

So that's the 'conventional' side of this EQ, and you should be able to see that it's a very versatile device in that regard, with wide-ranging bandwidth controls, more than enough gain/attenuation for most applications, and wide frequency ranges with some useful overlap. The presence of two separate channels means you can process two different sources simultaneously, work on stereo signals such as a drum or mix bus — it's definitely of a quality that I'd be happy using it in those applications — or perform detailed work on a single mono source, perhaps even using one channel for nixing nasties and the other for tone-shaping. For stereo work, there's no link function, so you do need to turn two controls every time, but that's easy enough given the detented knobs.

Then there's Crush, which really is a feature that's worth writing home about. Yes, it's a compressor and a saturator, and we've all used those. But here, it's somehow different, because it interacts with the EQ and being accessible right alongside it, it puts more options for coaxing the desired tone out of a recording to hand. It's somewhat like having an EQ and an exciter in the same control panel. Do you want to boost the high end? OK, try dialling it in with EQ. Not quite cutting it? OK, hit the Crush button and see what happens. Generally, you'll find that it tends to 'fill out' the band, making it... well, fatter or more shimmery, just as Drawmer say. You might find that you need to dial your EQ gain back a bit to compensate, as you often don't need to have a huge EQ boost to get the best result from Crush, and it can sometimes take a while to decide on precisely the best settings. But I can't state just how useful and addictive Crush can be.

Sticking with the top end, I found that this processor could really help to lift up all those engaging breathy details that can make a vocal sound close and intimate. On the right material, it could do that for a vocal in a relatively spacious stereo mix too, without too much collateral damage. And if you have a stereo mix or drum mix that's a bit lacklustre at the top end, it could inject a little more life there too. It was a similar story for the other bands: Crush acted as a sort of 'magic' button that filled things out and enabled me to emphasise a certain character, which I could not access with EQ alone, and I could then fine-tune it using the EQ band's gain control. I didn't always want Crush on, of course, but it's definitely a seductive effect, and while I've used plenty of exciters and saturators, this seems different from tools I've used before (and not least because it's to hand when EQ'ing).

Verdict

I'd certainly recommend that you give the 1971 a try if you have the chance. Even used as a conventional EQ, this thing is well specified and easy to tune in: clear labels, lots of control of the gain, bandwidth and frequency, easy recall, low noise, and in general a lovely, clean sound that means it works well on most instruments. Having the filters alongside the other four bands is really handy, of course. And then there's that special per-band Crush facility, which can be used subtly or with a heavy hand, as you prefer. I have to say that there really is nothing not to like here — the best I could come up with is the lack of a front-panel power button! — and I'm extremely reluctant to send this unit back... SSS

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Overloud TH-U Premium

Amp Modelling Plug-in

What do Overloud's Fluid IRs add to their flagship amp-modelling software?

PAUL WHITE

Overloud's amp and effects modelling package has evolved considerably since its launch in 2008 (we reviewed TH-1 back in *SOS* March 2009: https://sosm.ag/overloud-th1), with developments such as amp capturing included alongside a generous raft of amps, speakers, effects and routing options.

Overview

We're on the fourth generation of Overloud's amp modelling technology, which has been designed to capture the crucial non-linearities of overdriven amplifiers under real-world playing conditions. Currently there are 89 guitar amp models with a choice of 50 cabinets, and four bass amp models with two related cabinets. The amp capture system is based on component modelling, which is refined by taking multiple measurements of the real thing at different control settings to build up a more accurate picture of how a particular amp behaves. On top of that you get 84 pedal and rack effects, 18 different mic models, and the option of putting up to four mics on the selected cabinet.

Overloud TH-U Premium
$269

PROS
- Fluid IRs deliver a more 'real' sound and playing response.
- A huge choice of amps, speaker cabs and effects.

CONS
- Presets that include pitch-shifting sound less smooth when playing chords.

SUMMARY
An already excellent amp emulation just got even better.

An Amp Tweak facility lets the user modify the amp models by switching out valve types and/or emulating the effects of a Variac, which lowers the operating voltage. Users can also load their own speaker IRs, but the really big news — it's the main focus of this review — is that the speaker emulation now includes what Overloud call Fluid Convolution and ReSPiRe 2 technology. The software includes 20 emulations created by capturing complete rigs, and in all cases ReSPiRe 2 enables the user to shape the sound by changing and moving virtual microphones. In case you were wondering, ReSPiRe stands for Real Sound Pressure Response. This latest version (v1.4.26) sees all cabinets and IRs updated to Fluid IRs, while previous updates in version 1.4 have added pedal models including the Centuria (Klon Centaur) pedal, Tri-Chorus, Phase Stone, Reverse Delay and ModulaVerb. All these updates are free for owners of v1.4.

MIDI compatibility makes it possible to set up a MIDI floorboard using a simple click-and-learn system. Learned MIDI connections are then available in any preset, and as there's also a mobile iOS version of TH-U available, using it for live gigs as well as in the studio is a very practical proposition.

Fluidity

While conventional speaker IRs can capture both the frequency and time-domain responses of a speaker, they don't take into account the way that the speaker performs at different power levels and in different dynamic situations. A real speaker introduces harmonic distortion, something a conventional IR can't replicate, and this distortion changes under different drive conditions adding an important component to the sound. The harmonics generated by a guitar speaker are frequency dependent, with more harmonics are added to the lower frequencies than to the higher frequencies. Some systems use multiple IRs to try to capture changes that occur at different levels but again, as conventional IRs can't capture non-linearities, the important harmonic distortion information is lost.

While Overloud don't want to give away too much about their methodology, they do tell us that they use a proprietary format that embeds the additional data and that the process is related to Volterra Kernels, which some readers will know as a technique that's long been used by Acustica Audio in their Nebula and Acqua plug-ins to measure and replicate non-linear systems — it includes a lot of scary maths, but there are many published papers on the subject if you fancy some late-night reading! Overloud have developed their own slant on that technique, specifically optimised for speaker cabinet emulation. A SuperCabinet feature allows the blending of up to four separate IRs, each IR being focused on a specific frequency range using crossover-type filters. The result can then be exported as a new IR in the form of a WAV file. While each cabinet model, including SuperCabinet, now uses Fluid

IR, the original IRs are still available too, and user IRs can be converted into Fluid IRs using SuperCabinet. When you load a standard IR, TH-U analyses its response, then reconstructs its harmonic generation, dynamic response and its Z-curve.

Sound & Feel

The operational paradigm of TH-U holds few surprises and the fairly brief manual bears testament to the fact that most operations are intuitive. Items from a gear list, which is viewed by category, are placed in the centre of the screen to form your processing chain, which might typically be something like an overdrive pedal followed by an amp head followed by a cabinet, perhaps with some reverb or delay after the amp. There's also a comprehensive library of presets, again sorted into helpful categories, that show off the range of sounds available.

By default, the Flow IR versions of the cabinets are operational but you can go into the master settings menu and switch back to the old-school IRs if you want to hear the difference. A splitter can be dragged into the chain at any point so that you can, for example, run two amps and cabs in parallel, and when you open the details for the selected speaker cabinet, you can choose which mics to use, where to place them and how much to pan them.

For my tests, I used both a Strat and a Les Paul. Overloud's amp emulations have always been good and this latest generation behaves very much like the real thing, with an impressive dynamic response and a sense of 'springiness' that translates to a comfortable playing feel. The new Fluid-enabled cabinets really enhance that sense of touch responsiveness, and they also seem to emulate speaker compression, which feels very natural. Switching back to the older IRs for comparison revealed a significant difference in terms of dynamic response. The low end of the Fluid IRs also behaves in a more natural way when playing chords. For fans of the Klon Centaur, the Centuria overdrive is very detailed and responds much like the real thing, while the ability to tinker with the valves inside the amps can also yield worthwhile results.

I've tried more amp modellers than I'd care to count, and I have to say that very few of them react in such an organic way to playing dynamics. The clean sounds are warm and springy but the highs never become glassy. The gentle break-up blues tones work perfectly, and when you get to heavier rock, the sense of high-end aggression is balanced by a hefty punch and good note definition, again without the highs becoming irritatingly edgy.

Verdict

If you already own one of the TH-series packages, it's definitely worth upgrading to this latest version — and if you already have an earlier 1.4 version, an update won't cost you anything. There might not be many visual differences between this version and earlier ones, but with the new Fluid IRs, your ears will definitely discern a difference!

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Nektar Panorama CS12

Channel Strip Controller For Logic Pro

The Nektar Panorama CS12 offers hands-on control over Logic channels and any plug-in you can throw at it.

Nektar
Panorama CS12
$399

PROS
- Sensibly priced.
- Good use of colour in both the display and the controls.
- Comprehensive control section.
- The vast majority of third-party plug-ins work without any need to remap anything.

CONS
- No surround support.

SUMMARY
A truly comprehensive channel strip controller with excellent control mapping features.

PAUL WHITE

Since the decline of the tape machine/mixer paradigm, various companies have designed control surfaces to bring back a more tactile feel to working with DAW software. One of the early entries was the Mackie Control system, which could be expanded to create a console-sized surface with multiple banks of eight faders. Others went for a more compact solution providing transport controls, a single motorised fader and maybe a playhead position dial. Another alternative is a tablet linked to the host system to use as a touchscreen interface.

Nektar's Panorama CS12 supports Apple's Logic Pro and is an extremely sophisticated interpretation of the single-fader DAW channel strip controller. Those already familiar with Nektar will know that the company have a long history building controller keyboards that include comprehensive control functions, such as the Panorama P6. The CS12 is designed to access all Logic Pro channel strip parameters including the selection and control of plug-ins, for which the user has the choice of using the fader or the pan control to select and adjust parameters. Through the use of a generous amount of buttons and a Shift function, there is access to a number of Logic Pro transport, navigation and other functions — some 26 in total — and when it comes to plug-in control, both Logic's own plug-ins and third-party AU plug-ins

can be controlled. Essentially any plug-in that allows for parameter automation in Logic Pro will work right away.

Power for the CS12 is via the USB connection to the host computer (though there is also a 5V power port for use with an external supply if a 500mA USB port isn't available). Installation requires that Nektar's ControlCore software be installed and a few system settings to be made, after which the CS12 shows up in the Control Surfaces setup menu and is ready to go. macOS 11.5 or later is needed along with Logic Pro X 10.7.8 or above. ControlCore can be downloaded once you've set up a Nektar account and registered the CS12 serial number. The full manual is also available for download, though there is a quick-start guide in the box.

Layout

Because of its comprehensive control layout, the Panorama CS12 is necessarily much wider than most other single-fader controllers, such as the PreSonus FaderPort that I use in my own studio. The tough moulded case, which measures 390 x 185 x 65mm, has a USB port on the rear along with the power switch and a barrel connector 5V PSU socket. There's also a footswitch TRS jack (which can accommodate two footswitches via a Y-cable) and a point for fixing a security chain.

The case itself is fabricated in two parts with curved front and rear edges, with a smooth metallic grey paint finish and clear white legending for the dedicated controls and Shift functions. In the Transport and Mixer Channel sections, the buttons are all illuminated and show either their name or icon on the top. There are 13 rotary controls that use 'endless' analogue pots rather than encoders, as that offers better resolution; there's one for pan and 12 on the right with coloured backlighting to differentiate their function when controlling plug-ins.

The fader is on the far left of the case, inviting it to be operated using the left hand while the right hand looks after buttons and knobs. Both the motorised fader and the pots have a fine resolution capability of up to 14-bit. However, this fine resolution means it takes longer to scroll through options or values, so the magnifying glass button can be used to toggle fine resolution mode on or off to speed things up where necessary. By default the Fine mode is switched off, meaning the pots have 7-bit resolution and the fader 10-bit resolution.

A colour TFT display provides visual feedback and has four buttons below it to select pages where, for example, a plug-in's editable parameters won't all fit on one screen. Its top line shows the currently assigned parameter and fader assignment. When controlling instrument plug-ins, tabs at the bottom of the display provide a choice of mapping directly to Logic's Smart Controls or user settings.

"The Panorama CS12 is a very elegant yet affordable solution for those users who prefer hardware control to mousing around."

To the immediate right of the fader are the key mixer controls comprising a pan pot and buttons for Arm, Solo, Mute and Select. There's no dedicated surround panner, such as a joystick, for those working in multi-channel or binaural formats. Two six-segment VU meters sit above the Channel controls. The fader normally controls the selected channel or master volume but may also be used to control selected instrument or insert effect plug-in parameters by pressing the Select button so that it turns blue, then clicking on the plug-in parameter you wish to control. You can also switch the automation view on or off here, and use the fader to insert new automation points.

Below the display are three further rows of buttons for the key transport functions (which include Loop on/off, go to start, go to end, and Replace), Undo, Auto(mation), Metronome and Shift. The topmost row handles stepping through tracks, opening and closing plug-in windows, opening and closing the mixer window and switching the fader resolution. The Shift button accesses secondary functions such as setting markers, Redo, Quantize, setting locators and so on.

To the right of the display are three encoders for Zoom, Tempo and Data/Menu/Enter. Zoom normally adjusts horizontal zooming but used with Shift it controls vertical zooming. Seven different functions are available from the Tempo pot (Shift+Tempo scrolls through the choices) such as scrubbing, nudging, moving the playhead and clip gain adjustment. As it is easy to nudge the Tempo knob by accident, I suspect many users will change its function to playhead position.

Then we have the vertical row of 11 backlit buttons, where the button colours change to provide navigational hints. Two banks of eight slots can be selected using the topmost (9-16) button. These buttons determine how the four display buttons, the four RGB buttons at the bottom right and the set of 12 rotary pots are assigned.

The Channel and Send buttons get you to any available send controls for the selected track or relevant DAW channel. In Sends mode, only the pots corresponding to available sends are illuminated; for example, if your track has three sends, then only three pots light up. Using the buttons at the bottom of the display, you can navigate between level, pre/post-fader switching and send destination selection. Up to 12 aux sends per channel can be controlled, which is more than I've ever needed. These buttons can also be used for the direct selection of plug-in slots to keep the »

Things are pretty simple round the back, with just a quarter-inch footswitch input, a USB-C port, a power switch and an input for an external 5V power supply

Nektarine provides an automatic mapping option for virtually any plug-in.

» workflow as streamlined as possible. Volume and insert automation points can be written at the current play position by touching the fader cap.

That leaves the 12 knobs, which are used for accessing plug-in parameters. Their default backlight colour is light blue, after which colours follow the plug-in's colour. Four buttons below assist with colour customisation but can also be used in conjunction with the four buttons below the display to control up to eight switch settings for each control page. When controlling plug-ins, the plug-in parameters are shown in the display, set out in the same way as the pots that control them. The Page buttons are then used to access any parameters not shown on the first screen. This isn't quite as friendly as having a display window over each pot showing its current function but doing it that way would have added considerably to the manufacturing costs.

Working with automation is easy, with the Auto button toggling between Read and the selected Write mode for the track. You can either select the automation mode from the CS12 or in the usual way using a mouse/trackpad in the DAW. If you use the CS12 menu, you get additional modes such as Single Point, which allows you to write in new automation points at the current position.

Assignations

There are three methods of making your own control assignments, though having spent about an hour going through my own third-party plug-in collection, I've yet to come across one that wasn't already mapped. Quick Learn is a simple method of assigning a specific plug-in parameter to a physical pot. Select the plug-in, hold down Shift while moving the desired parameter, then move the control you want it to link to. You can also customise the control's colour if you wish.

Learn mode is the next level up and is a better choice if you need to assign multiple parameters. This time pressing Shift and Data gets you into a Learn mode where you can progress as in Quick Learn mode but you don't have to keep pressing the Shift button. Alternatively you can download the free Nektarine software, which can itself be run as a plug-in within Logic or run standalone. This allows additional functionality such as renaming parameters, drag and drop mapping and so on. Once you have saved the map file, it will always be available from within Logic.

User Impressions

The long-throw motorised fader and the clearly identified transport buttons make operating Logic very intuitive, though for some functions, such as setting loop points, I still find myself heading for the mouse as it seems more direct. In areas like this it really depends what you are used to. Overall I can't fault the functionality of the CS12 and virtually all the main functions are directly available, with only a few essentials requiring the Shift button. The support for the various automation modes is impressive, as is the ability to open and close Mixer and Smart Controls windows directly from the transport section. I also love the use of colour to tie together the various control functions, and I applaud the direct access to numerous send slots. When it comes to ergonomics, the control section is set out very logically but I would definitely switch the Tempo knob function to control playhead position instead.

Even though I was brought up using tape machines, mixers, outboard gear and patchbays, when it comes to controlling plug-ins, I still find myself heading for the mouse to make on-screen adjustments rather than using hardware, as to me it just feels more direct. Nevertheless I have to concede that for certain musical styles it makes sense to have hardware control over things like filter cutoff frequency and other effects that are 'played' in real time to record automation. In reality some users gravitate more towards hardware and others are happy to spend all their time using screen and mouse, but whichever way you lean, the access to plug-ins is very smoothly implemented here and knowing that the vast majority of third-party plug-ins will also work without needing to do any additional mapping is very reassuring. Furthermore, if you do want to create your own maps where all your 'hands-on' controls are in the optimum place, the free Nektarine software will allow you to do just that. While Nektarine is by no means an essential add-on, those with more ambitious needs will not be disappointed; the dedicated Nektarine user guide is well worth exploring.

In summary then, the Panorama CS12 is a very elegant yet affordable solution for those users who prefer hardware control to mousing around. The controls all feel comfortable and positive while the use of colour backlighting helps with navigation. If you're happy with a one-fader controller rather than a box with banks of individual channel faders, then the CS12 offers an extraordinary level of control for a very attractive price.

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The Neve 1073 is without doubt the most iconic channel strip of all time. Originally developed in 1970, as part of a custom console for Wessex Studios in London, it has been imitated countless times and reissued in a number of different formats — most recently in the form of the 1073SPX-D.

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Black Lion Audio Revolution EXP

Eight-channel A-D/D-A Converter

Audio interface modders Black Lion Audio enter the world of ADAT expanders.

MATT HOUGHTON

ADAT expanders add more I/O to compatible audio interfaces via an optical Lightpipe. There are lots to choose from, and the latest to jostle for our attention is Black Lion Audio's Revolution EXP, a 1U rackmount device that offers eight analogue line-level inputs and outputs (no mic preamps) and 24-bit A-D/D-A conversion.

Its two pairs of ADAT ports support up to 96kHz operation for all eight channels in both directions, or four channels at 176.4 or 192 kHz. All the analogue connections are balanced TRS jacks on the rear, where I prefer them (when hooking these up to rack gear or a patchbay you're not left with cables trailing from the front). Each input socket has a -10dBV switch. A pair of BNC connectors on the rear, cater for word-clock input and output. It's also possible to have a clock embedded in the ADAT signal, so you don't actually *need* a word-clock cable, and if you have an interface that lacks word clock that's particularly helpful. But word clock can be very useful, not least when sync'ing multiple devices. Also, worth noting is that Black Lion have pedigree when it comes to building low-jitter digital master clocks, and here they recommend using the word-clock connection where possible as it can often deliver more reliable sync.

Finally, a USB-C socket (reserved for firmware updates) is joined by a DC power inlet for the supplied switch-mode supply, a lightweight wall-wart type. I'd have preferred a locking connection of some sort, and ideally a PSU that terminated in an IEC socket for convenient connection to my rack PSU. But on the plus side, the external PSU keeps mains AC well away from the audio circuitry, and external supplies are easy to replace if ever they fail.

In terms of build quality more generally, the case is robust and nicely finished; flawless. The controls and meters couldn't be easier to use either. On the front, every input and output channel has its own eight-segment LED level meter, with input channels 1-4 and 5-8 in clusters to the left and output channels 1-4 and 5-8 to the right. It doesn't offer as fine resolution as, say, the meters on my Ferrofish A32's screen, but it's a large, clear, no-nonsense display. Plenty good enough to make the right judgements.

A push-turn encoder toggles the clock source and sample rate, while a minimalist screen and a few LEDs display the current status clearly. One more LED flashes red when sync'ing over a word-clock connection, settling to green when sync is established. While this doesn't occur for ADAT-only sync, that's by design — more on that below. Finally, the front panel on/off rocker switch — a feature all devices really should have today.

What this boils down to is a device that's intuitive to operate. What you see is what you get, and a quick glance tells you almost everything you need to know.

Impressions

To test the Revolution EXP I hooked it up to a couple of ADAT interfaces: a Ferrofish A32 converter (in turn connected to an RME MADIFace USB using optical MADI), which has both ADAT and word-clock connectivity; and an Avid MBox Studio, which supports ADAT but lacks word clock support.

Setting the Revolution's clock source to Macro-MMC (its internal clock), the Ferrofish to sync to the ADAT signal and the RME to sync to the Ferrofish's MADI clock, everything worked as expected: I could route sound from the Revolution EXP to the DAW and *vice versa*. With the same setup, I could slave the Revolution EXP to the Ferrofish. If setting the RME or Ferrofish as the clock master, though, while word-clock sync was fine, I was unable to get the BLA unit to sync to the ADAT clock. To check if it was an issue with the BLA unit or my setup, I tried it with the MBox Studio, and happily

The Revolution EXP includes both word-clock in and out as well as dual ADAT ports supporting S/MUX operation up to 192kHz for four channels, and a maximum of 96kHz for all eight channels.

the Revolution EXP sync'ed flawlessly over ADAT. Repeating the test with another device again revealed no problems.

I mentioned that when sync'ing via ADAT, you don't get the sync light to confirm that sync is in progress/established. BLA say they didn't want people to rely on an LED, as they've learned over years of modding interfaces that 'multiples' often sync but sub-optimally: a device set to 44.1kHz will line up on every other sample with another set to 88.2 sufficiently well to pass audio and show sync. So they prefer you to listen. This confused me at first, because the Ferrofish setup wasn't sync'ing over ADAT and I had no visual cue about the problem. But in general their approach makes sense.

As with BLA's audio interfaces, the main selling point in the company's literature is the sound. BLA's roots lie in modifying converters and audio interfaces to improve the analogue stages, through tweaks of the sort Sam Inglis described when reviewing their Revolution 2x2 interface (https://sosm.ag/black-lion-revolution-2x2). The technical specs are published on BLA's site, but BLA make it very clear that their aim is not simply to improve on-paper specs, but to judge the best configurations through painstaking listening tests. Consequently, they're keen to point out that they could have 'bettered' some specifications through different component choices but this would have sounded less good than what they settled on. For example, the dynamic range figures (112.837dB for the inputs, 110.175dB for the outputs) could be made more impressive, because they selected some components with a slightly higher noise floor than alternatives that, by implication, they felt compromised the subjective sound quality.

The sample rate and clock sync are easily set without any menu-diving required.

Clocking Off

With all that in mind, how does the Revolution EXP compare with other ADAT expanders? Well, it certainly sounds very good to me — a touch better than my Ferrofish A32 when listening critically over headphones (though it's hard to set up tests where the act of switching can't influence your judgement). Subjectively, I'd describe it as very clean, with a nice, extended top end. It's clearly very good.

Another factor one might judge an ADAT expander on is the feature set. Some might consider the Revolution EXP to be a little light in this respect: my Focusrite Clarett+ OctoPre, for example, not only has eight mic preamps but also caters for line and instrument signals, as well as offering the Air processing option and insert points, and it's not dissimilar in price. Another alternative is the Ferrofish Pulse 16, which provides twice the line-level I/O count with full S/MUX support and allows you to adjust the input gain and output level for each analogue channel to boot. In short, the competition is fierce, and inevitably for a US-made device, the price is a little more accessible in the US than the UK, while it's the other way around for European companies' devices like the Ferrofish. But for all that, I love the simplicity and no-nonsense approach of BLA's Revolution EXP. You want more line-level I/O to hook up some outboard to your DAW? Here it is. It's high quality, well built, sounds great and is very easy to operate — and there's an awful lot to be said for that!

Black Lion Audio
Revolution EXP
$799

PROS
- Sounds great
- Easy to use
- Clear metering

CONS
- Nothing of note.

SUMMARY
A line-level ADAT expander that's uncomplicated to set up, sounds great and can act as a master clock to boot.

$ $799.
E info@blacklionaudio.com
W www.blacklionaudio.com

AIR Studios Reverb's Mixer page contains eight stereo microphone signals and an Interactive Hall Guide, which shows the currently active mics and their positions in the hall.

Spitfire Audio AIR Studios Reverb

Reverb Plug-in

Spitfire's first effects plug-in captures the unique ambience of an iconic London studio.

DAVE STEWART

Thirty-three years after its acquisition by Beatles producer Sir George Martin, AIR Studios Lyndhurst remains a sought-after recording facility. While other London studios buckled under the weight of business rates hikes/the Home Recording Revolution/Brexit/Covid/Liz Truss (delete where inapplicable), AIR weathered the storm and emerged with a stellar client list (including Adele, Coldplay, Dua Lipa, Paul McCartney and Billie Eilish) and a string of award-winning scores for blockbusters such as *Gladiator*, *Interstellar*, *The Dark Knight*, *Inception*, the *Dune* movies, *Atonement*, *No Time To Die*, *Harry Potter And The Goblet Of Fire*, *Grand Budapest Hotel* and *All Quiet On The Western Front*.

The studio's popularity with film composers stems from the amazing acoustic of Lyndhurst Hall. Built in 1884 for congregational singing, this hexagonal, 5000-square-foot galleried space is blessed with a rich natural reverb that creates a glorious sonic halo around anything, from a guitar amp to a full symphony orchestra. AIR's former Chief Engineer Geoff Foster describes the hall as "a creative space in which you just want to perform", thus fulfilling George Martin's vision

of a musician-friendly environment within a spectacular recording location.

Over the years, two of Lyndhurst Hall's most frequent visitors have been Hans Zimmer (composer of seven of the aforementioned soundtracks) and UK company Spitfire Audio, who recorded their first orchestral sample library at AIR in 2008 and subsequently returned to capture dozens more. This longstanding relationship has led to a significant double first: AIR Studios have given Spitfire unprecedented access to sample the acoustics of their historic studio space, allowing the samplemeisters to create their debut effects plug-in, AIR Studios Reverb.

Overview

AIR Studios Reverb, or ASR as we'll call it, is a detailed convolution reverb plug-in that uses Spitfire's proprietary Virtual Positioning Technology to tailor the acoustics of the Lyndhurst Hall in any way you choose. This feature allows you to position your sound source anywhere in the hall while capturing the precise ambience of its acoustic spaces. You can mix between eight different stereo mic signals (see box) and adjust the source radiation between five orientations: you can also instantly switch the room from Spitfire's standard orchestral layout (players facing the control room) to Hans Zimmer's preferred orientation, which flips the whole setup by 180 degrees.

You can adjust ASR's five-band EQ by clicking and dragging the on-screen nodes.

To generate the 67,000 impulse responses (IRs) required to model a large three-dimensional space, Spitfire recorded full-spectrum sine sweeps played through 10 ATC SCM25A Pro active monitors: the speakers were set up in a row and systematically moved across the floor to collectively cover 28 positions, thus providing full coverage of the performing area.

Swivel-top stands were used to rotate the speakers between left, right, front and back orientations, with custom-made wooden trays laid on the stand tops enabling each speaker to face upwards. In a repetitive rigmarole lasting a full week, overlapping sine sweeps were recorded for nearly four minutes for each orientation, then repeated as a safety backup. Not my idea of a fun listening experience, but a member of the recording team described it as "more calming than sampling"!

Main Controls

ASR's Main Controls panel features Mixer, Source, Shape and EQ sections. The mixer houses eight stereo mic signals with individual fader, mute and solo controls along with an Interactive Hall Guide, a colour-coded representation of the microphones showing their position in the hall. Included on the visualiser are a pair of linked diamond-shaped icons ('pucks' in Spitfire-speak) which show which way the input sound source is facing.

The Source page allows you to change how the input signal radiates in the space. It contains five faders corresponding to the left, right, front, back and up orientations, with an additional sixth 'sub' fader dedicated to IRs recorded in the 20Hz to 200Hz range. A circular control rotates the sound source through 360 degrees, with an interactive graphic showing the coverage area.

Shape is a term used to denote digital manipulation of the reverb's envelope and length. Its three main controls are Direct Level (not to be confused with the wet/dry mix), which adds more transient response to the reverb entry and increases instruments' initial attack; Early Level, which increases the volume of the initial reflections which occur between 50 and 80 milliseconds; and Late Level, which controls later reflections that constitute the main, smooth-sounding part of the reverb.

The remaining Shape controls, Early Stretch and Tail Stretch, increase or decrease the length of the reflections. There are five stretch modes: Retro alters the IR's playback speed but also affects its frequency, while Granular creates a grainy digital effect by looping selected cycles of the IR. The remaining Elastique modes stretch the reverb tail without changing the pitch: its pristine High setting is slightly slower to take effect and more CPU intensive, Low is less detailed but acts faster, while Mid falls somewhere in between.

Meanwhile, the more familiar EQ page offers five bands of parametric EQ with frequency, gain and Q settings determined by a set of nodes you can drag around on screen. You can place the EQ before or »

Spitfire Audio AIR Studios Reverb

$349

PROS

- An utterly realistic, highly detailed 3D representation of an iconic studio's natural ambience.
- You can position the sound source anywhere in the hall using eight different stereo mic signals and five orientations.
- You can instantly switch between Spitfire Audio's and Hans Zimmer's orchestral layouts.
- Offers engineer-level control over the hall's physical characteristics and reverb reflections.
- Sounds beautiful.

CONS

- Not cheap, but this is a top-quality product.

SUMMARY

For the first time, a plug-in which replicates the legendary ambience of AIR Lyndhurst Hall. You can mix between eight stereo mic signals and position the source signal anywhere in the hall. Suitable for orchestral ensembles, solo instruments, percussion, voices and cinematic drums, and featuring Hans Zimmer's authentic AIR Studios layout, this epic-sounding reverb will bring instant size and sheen to your arrangements.

Microphone Array Options

Decca Tree: The traditional arrangement of three omnidirectional microphones. Placed high above the conductor's head, these give a strong representation of the hall while maintaining instrument focus.
Outriggers: Two omnidirectional microphones placed outside the Decca Tree to add extra stereo width.
Binaural: Neumann KU 100 Binaural Head (aka 'dummy head') placed on a stand to pick up the conductor's natural perspective.
Ambient: Two omnidirectional microphones placed towards the rear of the room capture Lyndhurst Hall's heavenly wafting room ambience.
Back: An alternative ambient pair located directly behind the Decca Tree to pick up the room's rear reflections.
Canopy: Microphones placed on the overhanging canopy pointing down at the floor.
Gallery Wide: Spaced mics placed high up on the two side galleries, good for more ambient, less directional longer reverbs.
Gallery Far: A pair of mics placed high up in the gallery above the control room, producing a similar lofty ambience to Gallery Wide.

» after the reverb, useful if you just want to (say) brighten the reverb without affecting the tone of the input signal.

Room Controls

A set of room controls at the top of the interface modify the hall's physical characteristics. Hall Layout switches the room configuration between Spitfire's preferred orchestral setup (Layout A) and Hans Zimmer's back-to-front seating plan (Layout B), which affects both microphone and player positions: this has a significant effect on the overall sound, and achieves with a single mouse click what would take many hours to set up in real life.

In order to reduce the excessively long reverb of the old Lyndhurst church hall, the studio installed a large, motorised hexagonal canopy suspended from the vaulted roof: this reduces the reverb time to just under three seconds, a manageable acoustic for instruments and voices.

In ASR you can adjust the height of the canopy between low (4.5 metres), medium (seven metres) and high (9.5 metres), the higher positions creating longer reverb decays. You can also switch between two different absorption materials used on the canopy: Melatech gives a slightly softer, less reverberant timbre, while Wood creates a harder and more reflective sound. The reverb time can be further reduced by enabling Gallery Damping, which adds sound absorption to the three galleries overlooking the hall.

Quick Controls

A simplified set of controls at the bottom of the interface provides quick access to important parameters: Input Gain governs the volume of the signal you send into the reverb, Mix controls the ratio of dry to wet signal and Output determines the overall volume of the plug-in. The vital pre-delay setting (which may be synced to your DAW's bpm) creates a sense of space by delaying the onset of the reverb, while Stereo Width and Mono control the input signal's stereo spread.

The ASR manual advises that making the source signal mono works well for solo instruments, while stereo is best for ensembles. Regarding the Mix setting, I'd also suggest that in most situations, using ASR as an insert requires a fair amount of dry signal in the mix, but when using it as a send it should be set to 100% wet.

The Source page allows you to change how the input signal radiates in the hall. The 'sub' fader is dedicated to IRs recorded in the 20Hz to 200Hz range: set it to 100% and you'll hear the hall's natural low-frequency response.

In Use

Rather than diving into the controls, I found it more fruitful to load some of ASR's factory presets to see what the plug-in is capable of. You can navigate the preset browser by clicking on pictures of instrument families, or by wading through an annoyingly non-alphabetical master list. Having chosen Celeste, and loaded Soniccouture's fine instrument of the same name, I was bowled over by hearing its close-miked acoustic transformed into a pristine, ethereal and enveloping large hall ambience. Magical.

I tried the same preset with Soniccouture's Grand Marimba (also great), Glockenspiel (exquisite) and Tubular Bells, which felt like being in Westminster Abbey on Coronation Day. The drily-recorded VSL Appassionata Strings were given a new lease of life by ASR's Expansive Strings preset, and using the Flute Solo preset with the similarly dry Flute 1 Vienna Instrument also gave great results. Ethereal Choir (which stretches the reverb to four seconds) sent the choristers straight up to heaven, while the all-purpose The Hall worked wonders on a range of solo instruments.

AIR Lyndhurst Studios, Hampstead, London. Built in 1884, the former Lyndhurst Road Congregational Church was acquired by producer George Martin in 1991 and converted into a state-of-the-art studio facility.

Though a longer reverb suits orchestral percussion, the Percussion Tight Mini Hall preset's 1.24-second IR works much better for ethnic and cinematic drums — I applied it to Tom Holkenborg's *Mad Max* Percussion and plaster fell off my music room ceiling. Used more subtly, the shortened reverb also adds a nice ambience to brass instruments and solo strings.

My verdict: as well as giving you the coveted AIR Lyndhurst epic orchestral sound straight out of the box, this lovely no-gimmicks reverb is a great tool for adding hall ambience to dry and close-miked instruments. Spitfire Audio and AIR Studios recording teams, take a bow.

Conclusion

You'll notice I haven't used the word 'immersive' yet: that's because the review copy was stereo only. Happily, Paul Thomson has now confirmed that Spitfire are working on a multi-out system that will be compatible with "as many variants of surround and immersive systems as possible". This will be available as a free update in a matter of months, so if you're itching to replicate Lyndhurst Hall's 3D ambience in an immersive audio environment, you hopefully won't have long to wait.

Though AIR Studios' longstanding success is a cause for celebration, nothing lasts forever: other great London studios such as BBC Maida Vale, Angel Studios and Abbey Road have all come within a whisker of being sold for residential development at various times. Whatever the future holds, I'm glad Spitfire have managed to capture the spirit of AIR Lyndhurst Hall in this excellent plug-in, which I strongly recommend to anyone looking for a wonderfully pure, big and beautiful reverb effect. ■■■

$ $349.
W www.spitfireaudio.com

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Audient ORIA

Audio Interface & Monitor Controller

Audient bring immersive audio a little closer to home with a dedicated USB audio interface.

PHIL WARD

Time was, not long ago, when considering a DAW interface it was the number inputs that caught the eye: does it have enough mic inputs to record a band with a drum kit? Now though, it's beginning to be the case that outputs are king. It's not about, 'can I record a band', it's more about, 'can I do an Atmos mix?' (Other multi-channel immersive formats are of course available.)

Philosophy

Up in the world of high-end professional Thunderbolt, HDX and Dante-connected interfaces, typical output count has long been more than sufficient for the 10 minimum required for Atmos monitoring (5.1.4 or 7.1.2), but down in the world of USB interfaces, where most of us reside, finding affordable hardware with enough outputs for Atmos, especially beyond 7.1.2 or 5.1.4, is a much bigger ask. And that's where the subject of this review, the Audient ORIA, comes in. The ORIA is a relatively affordable USB DAW interface designed specifically for Atmos mix duties. ORIA provides 16 surround monitor outputs (both analogue and AES digital) along with two independent stereo outputs and two stereo headphone outputs. In fact, the ORIA is so far away from a traditional USB interface and incorporates so many functions and facilities aimed at the mix, rather than the recording business of an interface, Audient cover it in the Monitor Controllers section of their website.

A good reason for this is that, while it can function as a traditional interface (which, for what it's worth, is how I think it will most commonly be used), it can also work in combination with another interface in a monitor control mode. In this mode, the ORIA is connected via ADAT I/O to an existing interface to provide up to 20 monitor outputs (and two headphone outputs) with all its monitor control functionality. If you're particularly wedded to using an existing interface, this is a perfectly practical option, but it seems to me that a better solution would be to use the ORIA as the DAW interface and then connect its ADAT input to a mic preamp/DAC ADAT output, or even an interface working in standalone mode. My Focusrite interface, for example, can be configured to function as a standalone mic preamp with an ADAT output, and that's what I'd use if I needed some more ORIA inputs.

Functionality

I'll begin by describing what the ORIA presents on its 1U rackmount front panel. Going from let to right, things kick off with a couple of combi XLR/Jack sockets that offer mic/line-level inputs. Alongside the input sockets are four configuration push-buttons that enable selection of 48V phantom power, input level and instrument impedance/sensitivity, independently for each input. Pressing one of the input selection buttons brings up its settings in the ORIA display, which coincidentally is the next feature along on the front panel. The display is both unusually generous in size for a 1U interface and usefully informative. The display reconfigures to show the information relevant to whatever is being adjusted (either from

Audient ORIA
$2699

PROS
- Thoughtful integration with Atmos routing and workflows.
- Versatile connectivity.
- Slick integration of Sonarworks SoundID Reference For Multichannel.
- Intuitive in use.
- Impeccable control app.

CONS
- None.

SUMMARY
With the OIRA Audient have not only brought dedicated Atmos mix facilities to the USB interface sector, they've created a really effective Atmos 'hub' device that potentially brings some much needed order to the Atmos muddle.

the front-panel controls or the associated ORIA macOS or Windows control app that I'll get to in a moment). To the right of the display is a large-ish rotary controller that, like the display, reconfigures depending on what's being adjusted or configured. Its default action is volume control of the ORIA surround outputs (by which I mean the 16 'Atmos' outputs) but if, for example, one of the two headphone outputs is selected, the rotary encoder switches to control that output's volume. Pushing and holding the encoder provides access to a settings menu with items and parameters selected by encoder rotation and confirmed through the push-button action. One small frustration I found with the encoder is that if the ORIA is placed directly on a desk rather than in a rack, it is a little too generous in its diameter to operate comfortably — the bottom edge comes too close to the desk surface. Maybe I just have thick fingers? The solution was simply to raise the ORIA up off the desk a little (which also coincidentally provided me with a convenient 'store and forget' slot underneath. Perfect for desk-cluttering paperwork...).

To the right of the rotary encoder is a square array of four configuration push-buttons. The top-left button is labelled Volume and it both selects volume control mode on the rotary encoder and provides a press-and-hold mute/unmute function. Alongside the Volume button is the Profile button. One major facility provided by the ORIA is the ability to import Sonarworks SoundID Reference For Multichannel speaker/room optimisation data. I'll describe how ORIA/Sonarworks integration works later in the review, but the action of the front-panel Profile button is to enable the selection of preset ORIA profiles that can include uploaded Sonarworks SoundID (or native ORIA EQ) data and ORIA output configurations. An example of the value of profiles arose pretty much straight away on installing the ORIA in my studio. My 5.1.4 Atmos monitoring comprises nine Dynaudio BM5 MkIIIs and one Dynaudio 18S subwoofer, while my stereo monitoring is via a pair of Neumann KH 150s. The Dynaudio Atmos system is optimised for the room using Sonarworks SoundID Reference For Multichannel, but the Neumann monitors are optimised using Neumann's MA 1 system, with optimisation parameters uploaded to the monitors. So switching from Atmos to stereo monitoring required the ORIA Sonarworks SoundID optimisation to be disabled, otherwise it would be applied to the Neumann monitors. ORIA profiles handled this perfectly. A press of the ORIA Profile button switched between the Atmos profile, including the Sonarworks optimisation, and a 'Stereo Flat' profile, while at the same time switching the ORIA output from multichannel to stereo. It was so neat I kept switching it just for fun, in a Homer Simpson kind of manner.

Bottom left on the push-button array is the ORIA Down Mix button. When the ORIA is used in conjunction with Dolby's external Atmos Renderer app, the Down Mix button is able to select between the downmix options configured in the Atmos workflow. Atmos downmix options might be 2.0 (conventional stereo) or 2.0 binaural, or any other multi-speaker format that needs checking. Perhaps, for example, you want to hear what your Atmos mix sounds like in Dolby ProLogic 5.1 format. At present, the ORIA Down Mix button can only be used with the external Dolby Atmos Renderer app and not with Atmos rendering integrated internally within a DAW. So, for example, it doesn't work with the relatively recently introduced Pro Tools integrated renderer, although Audient say they hope this will be able to change in the not too distant future. However, the lack of integration between the ORIA Down Mix button in internal Atmos renderers is in practice only a very minor detail. All the downmix functionality is still available within the internal renderer, it just can't yet be switched via the ORIA Down Mix button.

Last on the front-panel buttons is a Function button that can be configured to provide quick access to one of four ORIA output functions: Dim (default -10dB, but adjustable), Mono, Polarity and Reference Volume. The Mono and Polarity (reverses left channel and mixes to mono) work only on the ORIA stereo outputs, and the Reference Volume setting is fixed at -20dB although scheduled to become adjustable in a future version of the ORIA Control app. A talkback function that routes an ORIA mic input directly to a headphone output is also, say Audient, planned for the not too distant future.

Finally on the front panel are a couple of quarter-inch headphone outputs with a push-button above each one that switches the display content and rotary encoder action appropriately. The signal fed to the two headphone sockets can be configured in the ORIA control app to pick up channels in the Atmos Renderer. So for example, if you have binaural and 2.0 stereo downmixes bussed back to your mix session from the Atmos Renderer (this is what is meant by the term 're-render'), they can be assigned independently to the ORIA headphone outputs. This is incredibly useful because it means you can always check how an Atmos mix is working out when downmixed to binaural or plain stereo simply by slipping on a pair of headphones.

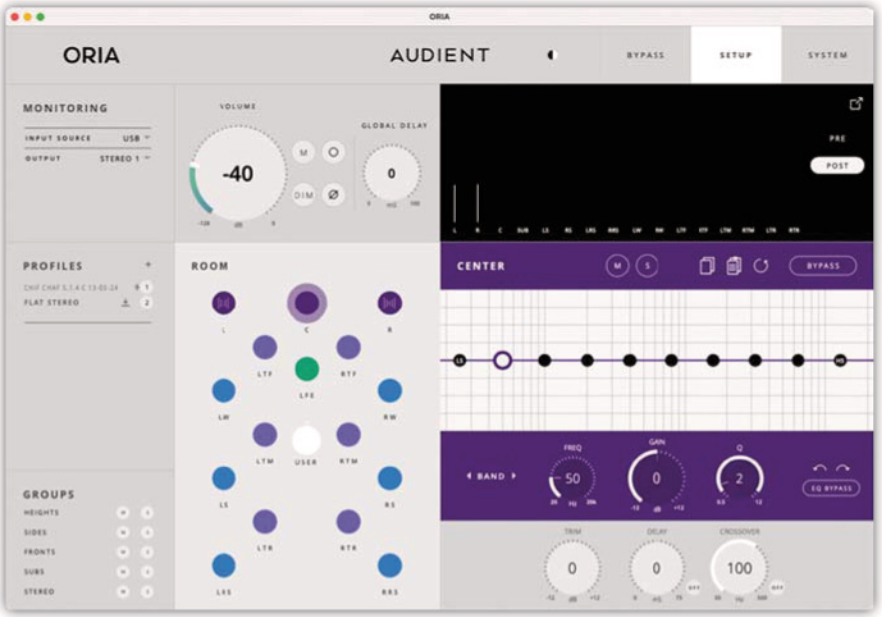
"It's genuinely exciting and inspiring in the way it dovetails so well with the creative opportunities and challenges that Atmos presents."

Round The Back

Introducing the subject of headphone channel assignments brings me neatly to the ORIA's connectivity on the rear panel. Starting on the right, the rear is dominated by 16 balanced jack monitor outputs. Rather than being labelled numerically, the monitor outputs are labelled with the acronyms

»

The ORIA's back panel features an IEC connector for the built-in power supply, a USB-C port, word-clock input and output connectors, an AES DB25 connector, ADAT I/O ports, 16 surround outputs and four stereo outputs.



Screen 1: The ORIA Control app setup screen with a Flat Stereo profile selected. Stereo output and no EQ.

» for Atmos channel identification, so for example, the Left Rear Surround output is labelled LRS and the Front Right Top output is labelled RTF. This is of immense help in terms of getting each monitor connected to the appropriate output, but it might cause a moment's head-scratching initially if an Atmos monitoring system is already in place — well it did for me. My existing 5.1.4 monitoring, based on a Focusrite interface with 10 line outputs, is connected following this scheme:

Output 1 > Left
Output 2 > Right
Output 3 > Centre
Output 4 > LFE (Subwoofer)
Output 5 > Left Surround (Ls)
Output 6 > Right Surround (Rs)
Output 7 > Left Top Front (LTF)
Output 8 > Right Top Front (RTF)
Output 9 > Left Top Rear (LTR)
Output 10 > Right Top Rear (RTR)

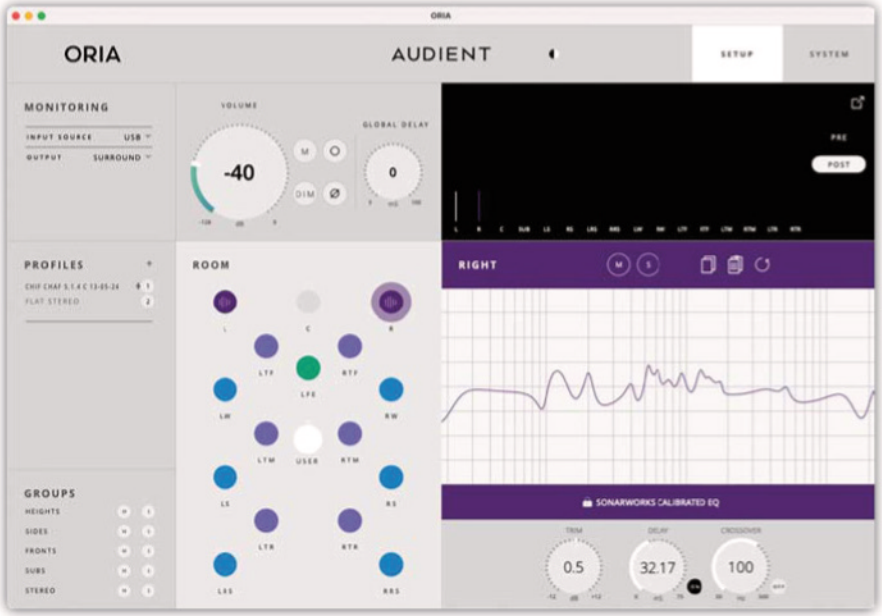
So the monitoring output channel routing defined in the I/O setup of my DAW reflects that scheme. However, with the ORIA providing 16 outputs (for 9.1.6), the channel numbers that the DAW sees (higher than channel 6) are not the same as my Focusrite scheme. This means that Atmos mix sessions that started with the Focusrite as the playback device have to have their output channels re-mapped for the ORIA. For example, the DAW output 7 (Left Top Front) has to become output 11 to feed the correct ORIA output. This isn't by any means a criticism of the ORIA, not least because once the session routing is mapped to the interface (and you've created an Atmos mix template with the appropriate output routing), you don't have to think about it again, but part of me would have liked channel number labels as well as channel acronyms printed on the ORIA rear panel.

Next along the ORIA rear panel are the ADAT in and out optical sockets I've already referred to, and above them is a multi-pin AES DB25 connector that provides 16 AES digital outputs for monitors with digital inputs. One of the bugbears of employing AES digital monitors in Atmos systems is that the paired daisy-chain cabling of the AES standard complicates studio cable runs. For example, the two numerically adjacent channels of one cable will, in an Atmos system, potentially be required to connect to monitors located a significant physical distance apart. However, the ORIA provides the functionality in its control app to redefine the AES channel assignments with regard to Atmos channel identification. This potentially makes cable routing in AES digital-based Atmos monitoring far more straightforward than otherwise might be the case. Just along from the AES connector is the ORIA's USB-C connection socket, a blanked Dante AoIP socket (Dante integration is scheduled, say Audient, for the end of June 2024) and, finally, BNC word-clock input and output sockets for clock sync use if peripheral digital hardware, such as an ADAT-connected mic preamp, is employed.

Control & Integration

So the ORIA is very well equipped with the connectivity required for both interface duties and Atmos monitoring, but it's perhaps the control app and integrated Sonarworks SoundID Reference For Multichannel that lifts it into a whole new category of relevance for Atmos mix work. I'll begin with Sonarworks integration.

If Sonarworks SoundID Reference For Multichannel is new to you, it's a hardware and software package comprising three elements: a calibrated measurement microphone, a room/monitor acoustic measurement/analysis »



Screen 2: The ORIA Control app setup screen with a 5.1.4 Surround profile selected. Multi-channel output with Sonarworks SoundID Reference EQ applied.

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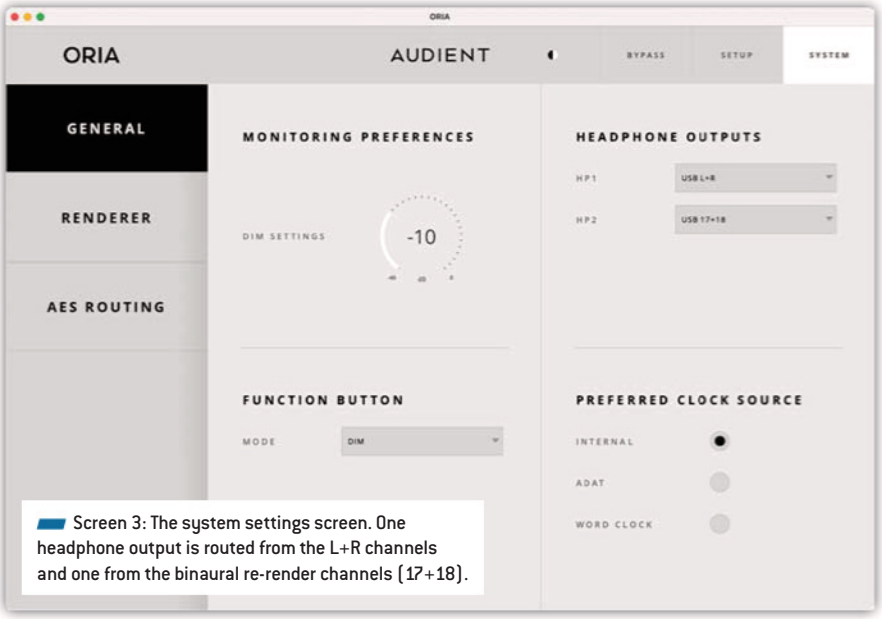


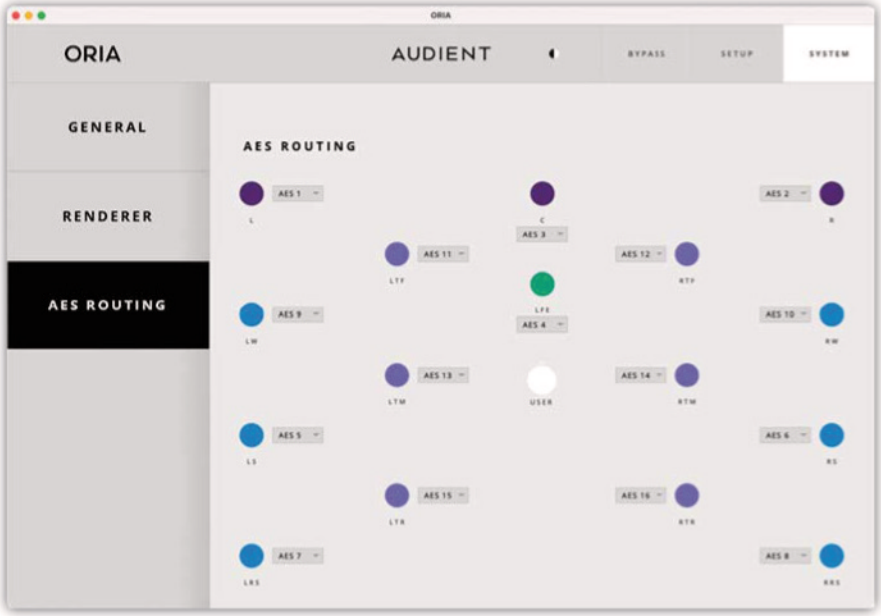
Screen 3: The system settings screen. One headphone output is routed from the L+R channels and one from the binaural re-render channels (17+18).

» app, and a front-end app that enables the results of the measurement and analysis to be applied to a monitoring system via either a DAW plug-in or through upload to monitoring hardware (either the monitors themselves or the monitor-driving interface). A Sonarworks SoundID Reference measurement microphone is packaged with the ORIA and it also comes with a 60-day Sonarworks SoundID Reference For Multichannel software licence. I fully described and reviewed Sonarworks SoundID Reference or Multichannel in a back in the April 2023 edition of the magazine, so I'll limit myself here to a recap.

The Sonarworks SoundID measurement app provides tools and procedures that enable a multi-channel monitoring system installed in a studio space to be acoustically measured. The measurement process is driven by the measurement app through on-screen prompts that make the process, despite the complexity that results from multiple monitors, reasonably fool-proof. The process does take a while, though, as Sonarworks SoundID requires measurements to be taken at 37 mic positions, with data taken at each one for every monitor individually. About 50 minutes is typical if you're practised at it, but the first time it'll probably take an hour at least from beginning to end. The result of the monitor and room measurement process is the creation of delay and frequency response optimisation parameters for each monitor that can then, via the Sonarworks SoundID front-end app, be either used in a monitor output bus plug-in or, with compatible monitoring hardware (such as the ORIA), uploaded and applied to the monitors directly. Uploading the SoundID parameters to hardware has significant advantages over their use in a bus plug-in: firstly, a plug-in that applies multiple EQ filters to up to 16 channels naturally demands a significant slug of processing power, and secondly, with the optimisation parameters located in the monitoring hardware, everything that leaves the DAW host computer can be optimised.

Uploading Sonarworks SoundID optimisation parameters to the ORIA is managed through pleasingly tight integration between the two apps. The Sonarworks SoundID measurement app checks at the commencement of the measurement process if the final destination for the optimisation data is an ORIA interface, and confirmation of that automatically creates a new blank profile in the ORIA app. Then, at the end of the measurement process, when it's time to upload the Sonarworks SoundID data set to the ORIA, that too happens with just a couple of mouse clicks. It's all very slick and well sorted.

Speaking of slick and well sorted, this brings me to the ORIA Control app itself. As well as duplicating all the front-panel functionality of the ORIA, the Control app offers access to a 10-band EQ (eight parametric, two shelf) for each output channel, a global delay function for lip-sync correction in AV mix environments, channel-specific delay adjustment, mute/solo functionality for both individual channels and groups (front channels, side channels, top channels, etc), and the previously mentioned profiles that enable groups of settings to be applied. Along with the Stereo and Surround profiles I described earlier, the combination of Sonarworks SoundID and ORIA profiles enables different Sonarworks target responses (Flat, Dolby, custom, etc) to be assigned and then easily compared. Finally on profiles, they are stored within the ORIA rather than the host computer, so once they are configured the ORIA can be disconnected from the host and the profiles selected from the ORIA front panel. This means that if an ORIA is



Screen 4: The ORIA Control app's AES routing setup screen, showing channel assignment drop-down menus for each monitor role.

ALTERNATIVES

If you're looking for a USB interface for Atmos beyond 7.1.2 (or 5.1.4), the ORIA has few competitors — the **Antelope Orion 32** being perhaps the only similar option. In the world of Thunderbolt or Dante, products such as the **Universal Audio Apollo 16**, **Apogee Symphony** or **Focusrite Red 16** for example offer a comparable I/O and functionality, but there's, quite literally, a price to pay.

used in monitor control mode, without a USB connection, previously configured profiles will still be available.

To illustrate some elements of the ORIA Control app functionality I've included a few screenshots. Screen 1 shows the Setup display with my 'Flat Stereo' profile loaded. The ORIA output defined at the top left is Stereo 1 (the ST1 rear-panel sockets) and neither Sonarworks SoundID data nor any user EQ is applied. Screen 2 shows the same Setup display, but this time my Atmos monitoring Sonarworks SoundID optimisation data is applied and the ORIA surround outputs are selected. It also shows the Sonarworks SoundID response optimisation applied to the currently selected (front right) output channel.

Screen 3 shows the ORIA Control app's System screen, which enables a few preferences to be set. Perhaps the most interesting options here are the headphone routing options. I have Headphone Socket 1 picking up the stereo Left and Right channels while Headphone Socket 2 picks up channels 17 and 18 — these are the channels where the binaural re-render is routed in my DAW Atmos template.

Finally, I've included Screen 4 for those who might want to use AES digital monitoring with ORIA. The screenshot illustrates the AES digital signal routing options I mentioned a few paragraphs up. A drop-down menu adjacent to the channel icons allows a specific AES channel to be assigned to each one.

Conclusion

In use, the ORIA is a delight. Bar one very small (and not particularly significant) bug in Sonarworks SoundID integration that I'm pretty confident will be fixed by the time anybody gets to read this, the ORIA works perfectly and intuitively. But more than that, it brings some much needed order to the Wild West world of Atmos mix routing and workflow — especially at the more cost-effective end of the Atmos universe. And as I sit here in my studio with an ORIA on the desk next to me, I can't currently come up with anything else I'd like it to do, or a manner in which I'd rather it behaved. I'm not quite sure how I'll go back to my previous Atmos interface solution.

I never usually get particularly excited about DAW interfaces. They've always seemed to me to constitute somewhat dry and functional chunks of hardware that have little bearing on the creative process. Nothing particularly to get the juices flowing. The ORIA, though, is different. It's genuinely exciting and inspiring in the way it dovetails so well with the creative opportunities and challenges that Atmos presents. Audient have absolutely nailed it with the ORIA.

$ $2699.99
W www.audient.com

Eventide Riptide

Overdrive & Uni-Vibe Pedal

Eventide's take on the legendary Uni-Vibe adds modern conveniences and a classy overdrive.

PAUL WHITE

Eventide's Riptide is a stereo digital pedal and the latest addition to their dot9 series. It combines the sound of the legendary Shin-ei Uni-Vibe with overdrive, which is a pairing often encountered in classic rock, perhaps most notably in recordings by Jimi Hendrix , Robin Trower and David Gilmour.

Overview

The two effects can be bypassed independently, each having its own footswitch, and they can be routed so that the Vibe comes before or after the overdrive. The Drive knob brings in what Eventide describe as a balanced distortion with a focus on dynamics and touch sensitivity, and there are two selectable drive voicings: Green, as denoted by the LED colour, has a mid boost, while Red offers a tighter, modern alternative that loses the mid boost. There are also two Vibe settings: Green is the classic Uni-Vibe, and Red a deeper, more phaser-like alternative. LED buttons above the two footswitches are used to switch voicings, as well as to indicate the current status by changing colour (red, green or unlit for bypass).

There are two ways to use the footswitches. In Perform mode, they bypass the effects in the usual way. Holding down the Vibe footswitch gets you into Preset mode, and in this case the Vibe footswitch steps around the presets and the Drive footswitch bypasses the whole pedal. There are user-selectable bypass options too, the choices being Buffered, Relay, DSP+FX and Kill Dry.

Structurally, the metal-cased pedal is very solidly built and the pedal comes complete with a 9V power adaptor and a USB cable. The controls are arranged as two rows of three knobs, the top three addressing Vibe, Speed and Intensity, with a button to the right to switch the order of the Vibe and Overdrive. Adjusting the Vibe knob from midway to full changes the Vibe's wet/dry mix so the effect can cover both the Chorus and Vibrato voicings of the original and anything in between. At lower settings the effect becomes less prominent and disappears altogether when the knob is turned fully anticlockwise. Below are three more controls, this time relating to the Drive section, with knobs for Drive, Tone and Level.

Connectivity has not been neglected. Small toggle switches select mono or stereo operation and guitar/line operating levels — you could use this pedal with synths or as a studio effect if you wanted, not just guitars. There's a single TRS jack for the input, so for most stereo sources you'd need to use a Y-cable, but there are two separate jacks for the outputs, and there's another one for external control.

Given the simplicity of this pedal and the onboard preset storage capability, I suspect many users won't need MIDI control unless they have a pedal switcher with MIDI capabilities, in which case being able to call up more presets, each with song-specific settings for the Vibe and Drive, could be useful. MIDI connection can be via the expression jack (using a converter cable) or via USB. MIDI Clock is also supported. The USB connection

Eventide Riptide
$299

PROS
- Authentic sound with the option to have the two sections in either order.
- Alternate voicings for Drive and Vibe.
- Supports MIDI and external pedals.
- Five presets available from the front panel or 127 over MIDI.
- Free Device Manager software for editing and settings.

CONS
- Separate input jacks for stereo would be nice.

SUMMARY
An authentic Uni-Vibe emulation, with the bonus of a good drive and some really useful connectivity.

also allows the pedal to communicate with Eventide's Device Manager software, of which more below.

There's also provision to connect an optional expression pedal to the TRS expression jack, this being able to control any combination of mapped parameters, allowing the user to morph smoothly between two different sets of control settings. Pedal parameters are saved separately for each preset. Alternatively, it is possible to use an optional footswitch via the expression jack to control tap tempo, and if a triple footswitch is used, that can also handle preset selection.

The Riptide has the ability to save and load five presets directly from the front panel, with up to 127 over MIDI. Eventide's free Device Manager software runs on Mac or Windows computers and can be used to create and save patches, as well as to provide access to software updates and system settings. A row of five LEDs indicate which preset is currently active. These same LEDs are also used to show the MIDI channel when setting up for MIDI operation (omni is set by default), selecting expression switch/pedal operation, and so on. There's also a Catch Up mode. When enabled, turning a knob after you've changed preset makes no difference until it passes the position of the current parameter value — a helpful touch that prevents abrupt changes.

The Device Manager software can be used as a preset editor/librarian, and also for making changes to system settings.

Good Vibes?

Important though the control functions are, what really matters is how the pedal sounds, and on that score I have to say that I'm impressed. I expected Eventide to get the Vibe sound right — and they have executed it beautifully — but the overdrive section is also good. In the more transparent Red overdrive voicing, the result is not unlike that from the Fulltone OCD that I use myself, whereas in the Green 'mid hump' position there's a definite Tube Screamer flavour to the sound. In either voicing, note definition remains good even at maximum drive, which is enough to get you into classic rock territory. At lower drives you get a bluesy break-up, and in all cases the touch responsiveness is good, allowing the player to control the intensity of the drive by picking dynamics or the guitar's volume control. In other words, even when you're not using it as a Vibe, the Riptide puts two classic drive flavours on your pedalboard.

Switching to Vibe, the distinctive throb of the modulation is replicated perfectly but there's also plenty of scope for dialling back the severity of the effect if full-on Vibe seems just a little too much. Both the Vibe Chorus and Vibrato voicings come across very authentically in Green mode, but switch to Red and you get more of a 'Vibe meets phaser' sound but still with the familiar Vibe pulsing modulation. Combining the Drive and Vibe also delivers very pleasing results. Putting the Vibe before a moderate amount of Drive gets you that coveted Hendrix flavouring while putting the Vibe after the drive delivers a much stronger tonal modulation as the Vibe then has a lot more in the way of harmonics to chew on. Both are musically valid sounds.

In summary then, the Riptide offers a very practical way to add both overdrive and a Vibe emulation to your pedalboard, all for a cost that is probably below that of buying two separate pedals, with the bonus that you also capitalise on pedalboard space. For the more sophisticated user, the MIDI functionality and pedal/switch functions will be a welcome addition, while the free Device Manager makes creating and saving presets easy and also simplifies access to system settings. This is one pedal that I'll be sorry to see go back. ■■■

The Riptide can handle instrument- or line-level sources, and there's plenty of connectivity on offer, with stereo signals catered for by the single TRS jack input.

$ $299
W https://eventideaudio.com

Genelec 8381A

Preview: Point-source Main Monitors

We take a deep dive into the innovative tech behind Genelec's new flagship loudspeakers.

SAM INGLIS

Every so often, a product comes along that seems deserving of *SOS* coverage, but can't easily form the subject of a conventional review. The new Genelec 8381A is one such example. Standing a metre and a half tall and weighing in at a colossal 235kg, it's not something that could easily be shipped to a reviewer or set up in a test space. And at around £60k per pair, it's not something anyone sane is going to buy on the strength of a review alone. Nevertheless, the 8381A is a product that breaks new ground in several ways, so even if it's not practical to put it under the usual review lens, it does seem worth investigating — hence this 'tech preview', which I've compiled by grilling Genelec's Andy Bensley, Jussi Väisänen, Juha Holm and Aki Mäkivirta.

Out Of The Walls

The Genelec 8381A is a main monitor, and thus belongs to an entire product category that doesn't see much coverage in *SOS*. That's partly because few of us have the budget or the need for such things, but also partly because main monitors are typically designed to be built into the fabric of the studio. Genelec's own 1236A three-way speaker, for example, will usually be found soffit-mounted within the front wall of a control room, rather than free-standing.

"There is quite a bit of physical separation between the bass drivers in the two units, and this is entirely deliberate."

In-wall mounting is often the absolute best option from an acoustic point of view, but it only makes sense as part of a full-on, permanent studio build. This requires a huge investment of money, materials and time that will have to be written off if you ever move to a different studio. Genelec, along with other manufacturers such as Amphion and Ocean Way, have identified

a growing demand for systems of the same power and quality that can be set up in less conventional or controlled spaces, and follow peripatetic high-end producers on their travels. The result is the 8381A, billed as the world's first point-source main monitor.

Point To Point

What does "point source" mean in this context, and why is it significant? Well, room acoustics aren't the only reason why main monitors are traditionally mounted in the front wall of the studio. Another reason is because they need to be several metres from the listening position in order to sound right. The sheer size of the individual drivers in a conventional main monitor means that their respective contributions to the sound don't integrate properly until you get quite far away from them. And this is one reason why studios also have nearfield monitors, because you can't approximate nearfield listening by simply moving closer to your main monitors and turning them down.

Coaxial drivers have been a major focus of development at Genelec in recent years, leading to the highly successful The Ones series of monitors. The idea behind a coaxial system is to have the tweeter mounted in the centre of the woofer, so that they appear to the listener as a single full-range device, regardless of listening distance. All the Ones are three-way models, with the bass drivers also arranged in such a way as to make the entire speaker behave, as much as possible, as a point source.

The largest of the Ones, the 8361A, has apparently been a big success for Genelec. In conjunction with the W371 adaptive woofer system, it's often able to perform the role that was mentioned above, offering something approaching main monitor performance in a format that can be used free-standing, at nearfield distances if necessary. For the company's 45th anniversary, they decided to create a new flagship that would take this concept to several new levels.

Each 8381A is driven by a pair of RAM 81 amplifier units.

The 8381A is thus not a three- or four-way but a five-way system, with each speaker comprising two main components: the four-way top/mid driver and the "adaptive double low woofer system". Unlike the W371, the 8381A sub is not optional, but an integral part of the system. And whereas standard subwoofers are often positioned elsewhere in the room, the 8361A's woofer system is closely integrated with the mid/top part and forms a stand for it. As with other Genelec models, it's an active system, but in this case the amps are separate units rather than being built into the speakers.

Like most of Genelec's current selection of active monitors, the 8381A has built-in DSP, and it forms part of their Smart Active Monitor range. This is significant for two reasons. Firstly, Genelec's GLM software can be used to measure and correct the frequency response at the listening position. And secondly, it can freely be integrated with other SAM models. You could, for example, have 8381As as your main studio monitors, and augment them with different The Ones models as surrounds and overheads to build up an Atmos rig. GLM will handle all the necessary alignment and correction functionality, and even offers speaker switching and monitor control courtesy of the new 9320A SAM Reference Controller. This stands in sharp contrast with the 'traditional' way of doing things, where the performance of a main monitor is largely defined by the architecture of the room and the wall within which it's mounted, and where any DSP correction would typically be performed by an external unit, requiring laborious and complex setup from an expert.

Big Beasts

In the all-black finish (white is an option), the 8381As bear a passing resemblance to the obelisk from *2001: A Space Odyssey*, and give off the slight air of having their own gravitational field that can bend light around them. You'd definitely want to check the load-bearing capability of your floor before ordering a pair, and have clear plans in place for getting them »

» up any stairs. As well as the mid/top section, the woofers and the amps, the system also includes a choice of angled wedges to ensure that the first of these elements targets the correct height at the listening position.

Treble and upper midrange are catered for through a relatively conventional coaxial driver, with the tweeter sitting in the centre of the larger midrange driver and the crossover at 1.8kHz. Below 500Hz, the upper midrange driver hands over to what is definitely the 8381A's most distinctive feature: an array of four five-inch drivers arranged in a square formation around the edges of a sophisticated waveguide that houses the tweeter/mid driver. Genelec's research showed that it would not be possible to achieve the necessary sound pressure levels in this frequency range by simply making the main coaxial driver larger or driving it harder, and that attempting to do so would introduce unworkable levels of displacement and hence distortion. At the same time, they felt it was important to retain the impression of a point source within this range, rather than handing over to a subwoofer.

The result was this novel driver array, with the four drivers equally spaced around the waveguide and housed in domed enclosures. Both the ridged waveguide and these domes are the result of laborious computer modelling aimed at minimising interference between the coaxial driver and this midrange 'filler' array. There was no off-the-shelf driver or enclosure that would work in this role, so everything had to be developed and manufactured in-house.

Making Adaptations

In the bass region, by contrast, the 8381A turns the 'point source' philosophy on its head. The upper unit that contains the coaxial mid/tweeter and the four-way 'filler' array also houses a forward-facing 15-inch driver, while the woofer system that sits beneath it is home to two side-firing 15-inch drivers, and a bass reflex structure that vents to the rear. There is therefore quite a bit of physical separation between the bass drivers in the two units, and this is entirely deliberate. In the frequency range where wavelengths are measured in metres, the human ear cannot readily detect which direction sound is coming from; there would be relatively little benefit in creating a monitor that behaved like a true point source down to 30Hz even if it was possible. But it's precisely in this frequency range where room modes inflict damage.

The physical separation between the front-firing 15-inch bass driver and the side-firing woofers is entirely deliberate and allows the 8381A's response to be tailored to room acoustics simply by varying the crossover frequencies.

Hence, what's adaptive about the "adaptive woofer system" is the crossover frequencies. Because the low-mid array, the front-firing bass driver and the side-firing woofers all occupy significantly different positions within the room, they all excite the space in different ways. So, rather than attempt to generate all the low-frequency sound at a single point and use EQ to compensate for the vagaries of the room, the GLM system intelligently varies the crossover frequencies to try to minimise cancellations at the listening position. The crossover between the low-mid array and the front-facing driver can operate between 250 and 150 Hz, while the sub unit can take over anywhere between 100 and 50 Hz, depending on what GLM deems most appropriate for the space.

The end result is a main monitor that, in principle, behaves like a sort of supersized coaxial nearfield above 200Hz or so, but which works with the room acoustics to deliver the most balanced low-frequency response possible in the space. As I said at the start of this article, I'm not writing a review here: I haven't set up these speakers in a known space, been through the calibration procedure myself or compared them with other competing models. I have heard them, though, and been suitably impressed. The sort of environment in which these speakers are typically demo'ed is rarely acoustically perfect, especially at the low end, and a speaker that can put out high SPLs at frequencies down to 20Hz would often cause more problems than it solves — especially at the kind of listening distance that is usually required with main monitors.

To my ears, the 8381As sounded focused through the midrange even close up, and even in demo spaces that were far from perfect, the bass response sounded even and true down to very low frequencies. Ideally, of course, we'd all be mixing and mastering in purpose-built rooms where all the modal resonances have been eliminated by careful design, but in the real world, that's often impossible even at the top level. The 8381As don't eliminate the faults of the room, and I'm sure Genelec would never claim otherwise, but they have the potential to deliver a true main monitor listening experience in environments where that might not otherwise have been possible.

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AIAIAI Unit-4

Active Monitors

Danish newcomers AIAIAI make monitoring portable with these compact, battery-powered wireless speakers.

PHIL WARD

As a product of the 1980s UK hi-fi speaker industry, I've been conditioned to believe that British speaker design leads the world. And for a while, in the 1960 and 1970s, there was some truth to that. Companies such as Quad, Tannoy, Celestion, KEF, Wharfedale and Goodmans (and the BBC) begat an entire industry of speaker designers and manufacturers for whom the cutting edge of electro-acoustic product design and engineering was currency. But a look North towards Scandinavia, and Denmark in particular, reveals some not insignificant competition. I write this because the Kombo portable PA I reviewed recently is the product of a Danish audio start-up, and now this review of the AIAIAI Unit-4 falls into the same 'Danish start-up' category. Denmark has produced more than its fair share of electro-acoustic products over the decades (think Dynaudio, DALI and Bang & Olufsen for a start) and is currently, I think, on a roll.

As far as I'm aware there's nothing else aimed at nearfield monitoring quite like the Unit-4. It's a portable and compact monitoring solution designed for use on desktops and in small studio spaces. Not particularly unusual — but the Unit-4 is also optionally battery powered, with a claimed 20-hour battery life, and wireless, and those two features differentiate it almost entirely from any competition.

Little Boxes

The Unit-4 is really pretty small and decidedly cute. I fell for its looks and feel almost instantly. It comprises an entirely plastic enclosure of roughly 3 litres internal volume, fitted with a nominally 80mm (3.5-inch) bass/mid driver and a 19mm-dome tweeter. Amplification comprises Class-D modules for bass driver and tweeter, rated at 30 Watts continuously but with an 80 Watt peak capability.

The bass/mid driver diaphragm is of coated paper, the tweeter dome is fabric, and the drivers appear conventional in terms of design and engineering. The bass driver roll-surround is generously dimensioned for the diaphragm size, which suggests that the driver is designed to offer reasonably extended excursion — which may go some way to compensate for its limited diaphragm area. The Unit-4 is a reflex-loaded monitor with an extremely long and tortuously routed port of modest cross-sectional area that terminates in a semi-flared exit on the back panel of the cabinet. Long reflex ports are prone to organ-pipe resonances, and ports without much cross-sectional area tend to become non-linear and compress at modest volume levels because they simply can't pass enough non-turbulent airflow. I'd expect both issues to be apparent in this design, but we'll see when I get to some acoustic measurements later.

The enclosure is usefully wedge-shaped to enable desktop landscape installation with a suitable upward angle, and it's manufactured from post-consumer plastic. AIAIAI make much in their promotional copy about the Unit-4 being easy to dismantle (with screws and no glue) and repairable, and they also offer a full range of spare parts purchasable from the website, along with repair guides and even trade-in options. These are a very positive developments to my mind, and I hope other monitor manufacturers will take note.

Some optional extras include a very smart carry case that takes a pair of

AIAIAI Unit-4
$800

PROS
- Effective and capable compact monitoring.
- Battery powered and highly portable.
- Optional wireless connectivity.
- Cool and stylish looks.
- Uses recycled plastics and is fully repairable.

CONS
- Slightly challenged by high volume levels.

SUMMARY
The Unit-4 is a very impressive first monitor effort from AIAIAI. It offers a unique set of features that ought to attract a good number of customers.

Unit-4s with space for cables and accessories, a recycled-silicone bumper cover that offers extra protection from the trials of portability, and recycled-silicone angled desk wedges that decouple the monitors and help with their alignment when desk mounted.

An interesting and attractive feature is that the bass/mid driver is surrounded by a ring of white LEDs that displays battery charge state, volume and wireless connection status. And the front panel is finished with a magnetically attached perforated steel grille that protects the drivers and looks pretty cool.

As far as signal connection options are concerned, the Unit-4 offers balanced and unbalanced analogue inputs on 6.3mm and 3.5mm jack sockets respectively, and wireless connection via either Bluetooth or AIAIAI's proprietary W+ Link protocol. The W+ Link, say AIAIAI, has advantages over Bluetooth in offering both lossless audio and low latency. Having said that, the 16ms latency specified is well within the range that might negatively affect overdub recording (although it's likely that you'd use a wired connection in that scenario). W+ Link works with the Unit-4 via an AIAIAI transmitter dongle, called an X02, that can be either connected to a USB-C port on your computer or can digitise and transmit analogue audio independently through its own stereo 3.5mm jack line level input. When connected to a host computer, the X02 appears as a USB-powered output device, but when used to transmit from its analogue input it must be separately powered — a USB smartphone charger will do the trick. Pairing Unit-4s to the X02 transmitter, either in stereo pairs or singly, is simply a matter of putting the speakers in wireless pairing mode by pressing and holding the rear panel W+ button and powering-up the X02 transmitter. Bluetooth pairing is similarly simple, and even if you don't use Bluetooth for audio, it's worth taking the time, because there's a Unit-4 smartphone app that provides some enhanced control and configuration functions that aren't available otherwise: EQ controls and presets, LED display brightness and firmware updates.

The rear panel houses the wired audio input connections, as well as buttons for engaging the various wireless modes and selecting left/right/mono speaker operation.

Pushing Buttons

The top panel of the monitor offers three push buttons. The left and right buttons enable volume control, and the middle one provides multiple functions such as power off/on, mute and battery check. If two monitors are wirelessly connected as a stereo pair, pressing a button on one of them also controls the other. The previously mentioned LED display ring illuminates in a variety of patterns and modes to provide feedback on button presses and the resulting monitor state, though I found it somewhat difficult to remember and interpret the meaning of the display. Hopefully familiarity would breed more immediate understanding. I've already mentioned the rear-panel wireless mode button that provides pairing and switches between Bluetooth and W+ modes, and there's also a button that assigns left, right or summed mono roles to a monitor.

On the battery power front, each Unit-4 monitor is supplied with its own mains power supply/battery charger device, and the monitors can be used with this connected or not. The battery is installed internally within the monitors and not accessible. I didn't formally check the claimed 20-hour battery life, but I had the Unit-4s working occasionally on my desk for quite a few days without connection to power so I've no reason to question it.

Measuring Up

Taking a Unit-4 to my acoustic measuring space and launching FuzzMeasure, I first investigated its axial frequency »

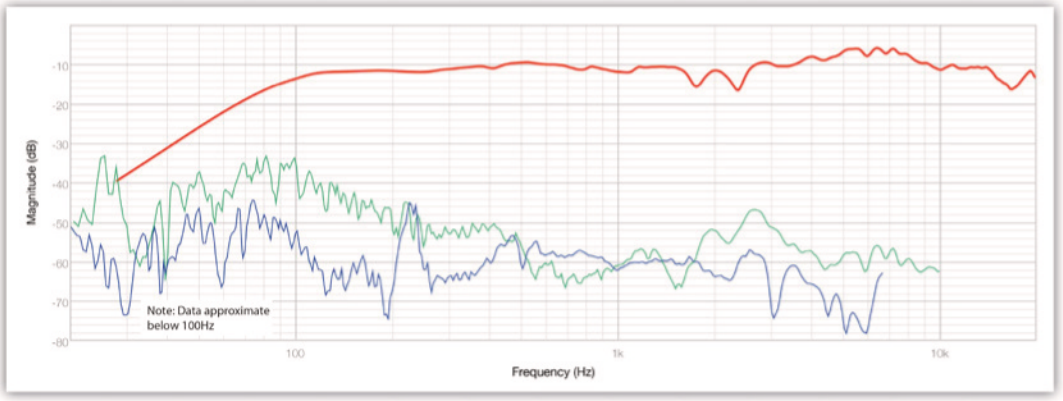


Diagram 1: The Unit-4's on-axis response, measured at 90dB at 1m. Second- and third-harmonic distortion is shown in green and blue, respectively.

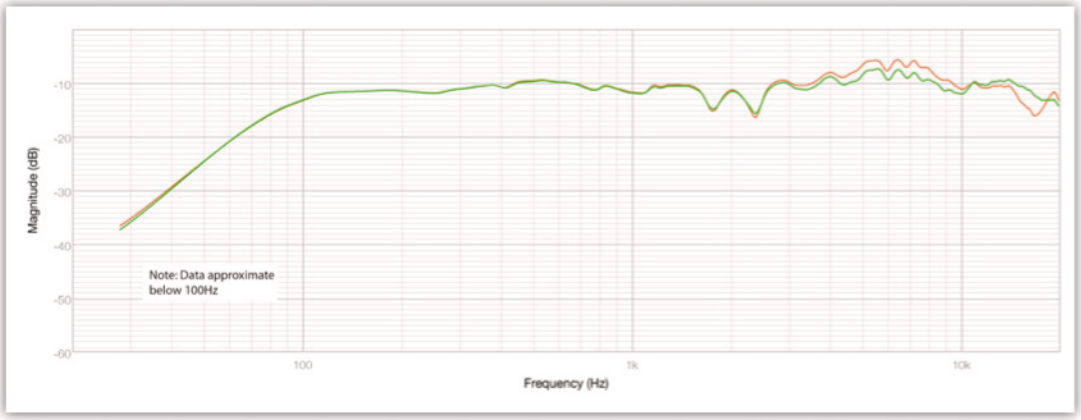


Diagram 2: Frequency response measured with grille attached (red trace) and removed (green).

» response and harmonic distortion, and the result of that can be seen in Diagram 1. The frequency response curve is decently flat, with just a couple of wobbles through the 3kHz crossover region. In the tweeter band, though, there's a conspicuous lift in level between around 3.5 and 9 kHz, and my feeling is that this will result a noticeable emphasis on cymbals, sibilance and generally on the high-frequency elements of programme material. Unit-4 distortion performance, shown in the green and red overlays on Diagram 1, is pretty reasonable, with the third harmonic typically around 40 to 50 dB down in the critical mid band. Considering the sound pressure level for the measurement was 90dB (at 1m), which is relatively loud for such a compact monitor, this is a good performance. The only slight fly in the cliché is the unexplained jump in the third harmonic at just over 200Hz, although I think it's unlikely to cause a specifically audible problem.

I mentioned the Unit-4's grille earlier, and wondered whether it would affect the tweeter frequency response. And the answer, illustrated in Diagram 2, is that it does. Diagram 2 shows the Unit-4 axial frequency response with (red curve) and without (green curve) the grille fitted, and it reveals that the high-frequency lift I mentioned in the previous paragraph is significantly reduced with the grille removed. This difference between grille fitted and removed is audible, and I did all of my critical listening *sans* grille. Now, you might be wondering if I've got the curves the wrong way around — perhaps you wouldn't intuitively imagine that the tweeter level would reduce with the grille removed. What I think is going on is that fitting the grille increases the radiation impedance of the air in the vicinity of the tweeter dome, effectively increasing its efficiency.

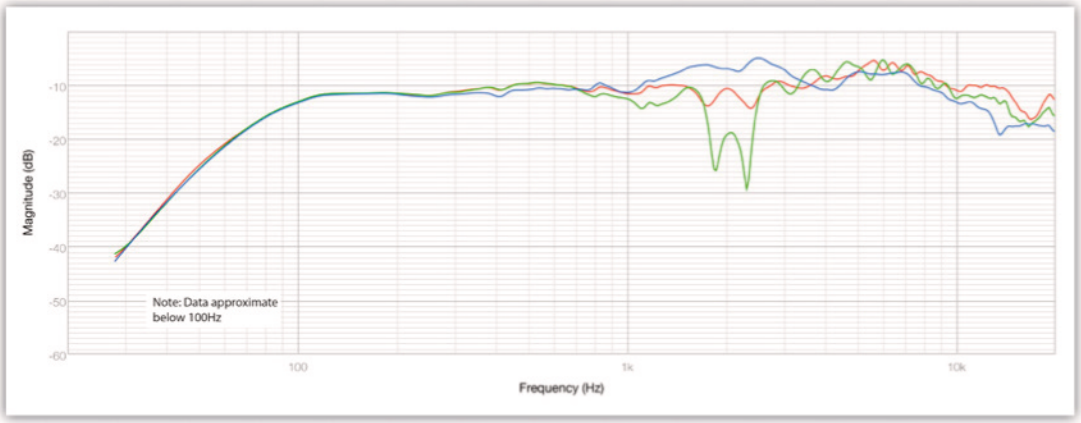
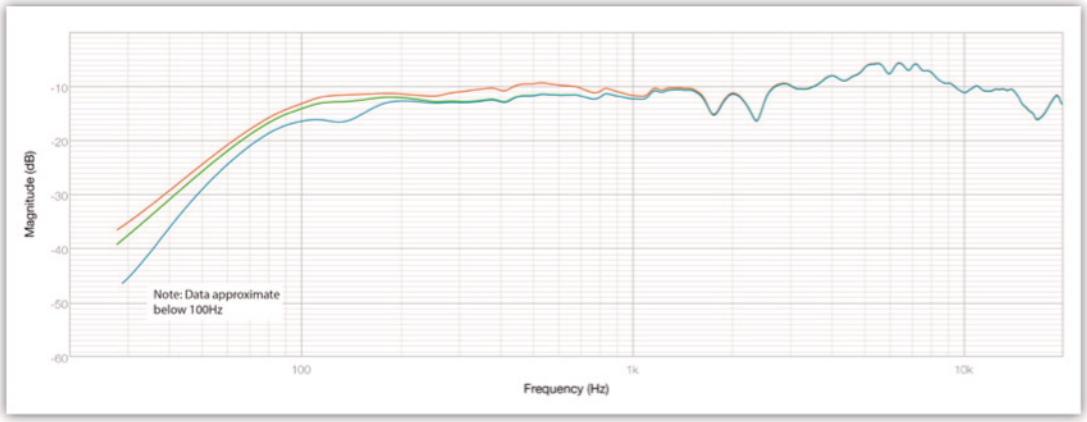


Diagram 3: Comparing the on-axis response (red) with responses 15 degrees off-axis, on the HF side (green) and LF side (blue).

Moving on, Diagram 3 shows the ±15-degree vertical (or horizontal if a Unit-4 is used in landscape mode) off-axis frequency response overlaid on the axial. The off-axis responses show the typical variation found in multi-way speakers as the driver phase changes through the region where their outputs overlap. However, if I were using a pair of Unit-4s in landscape mode on a desk I'd try to ensure that they were angled inwards towards the listening position, because off-axis in either direction results in some potentially significant response variations. Portrait orientation with the speakers raised to ear height or angled back would probably a better option — the Unit-4s consume less desk space that way too.

Diagram 4 illustrates some of the EQ presets available through the Unit-4 app. I've chosen to illustrate just the -3dB and -12dB desk options, and those numbers seem to refer to the maximum attenuation that is produced; at 100Hz, the -12dB curve actually only produces around -6dB attenuation. Along with the desk EQ presets, the Unit-4 app also offers a five-band graphic EQ module. Low bass, bass, mid, upper-mid and high-frequency bands can each be adjusted between -12dB and +3dB. Without wishing to give the game away on what I thought of the Unit-4 when it came to listening, I found the EQ and presets generally useful in terms of tweaking the subjective tonal balance to suit a variety of desk-mount installations — although I can't imagine anybody needing to use the LF -12dB preset. The LF attenuation of the desk presets is also valuable in enabling the Unit-4 to play louder, as it effectively increases amplifier, driver and reflex port headroom.

Diagram 5 shows the close mic frequency response of the Unit-4 reflex port output, measured at two drive levels (equivalent to 80dB and 90dB at 1m) and three things are apparent. Firstly, a port organ-pipe resonance was confirmed. In fact the fundamental and second harmonic of the resonance are easily identifiable at just over 400Hz and 800Hz, and I'd expect the port resonance to introduce some subtly audible coloration. The second phenomenon



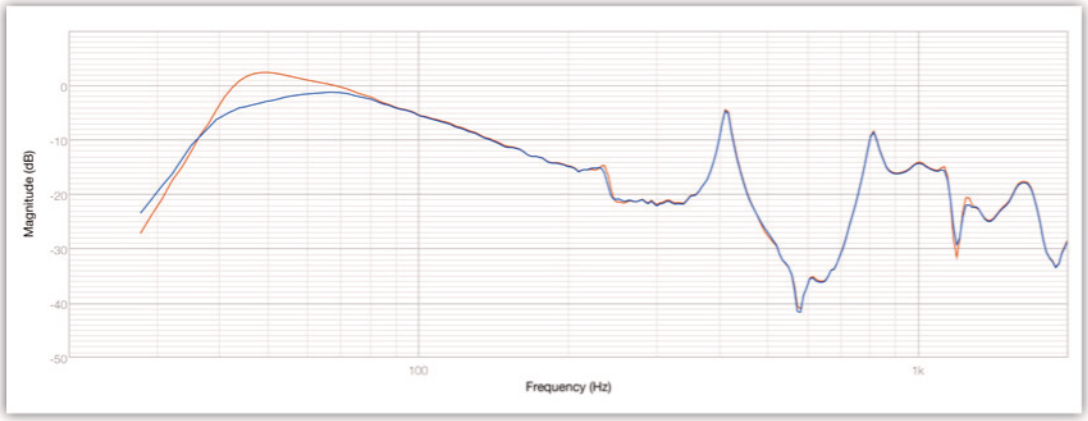


Diagram 4: The Unit-4's response with EQ set to neutral (red), and with -3dB and -12dB desk EQ options engaged (green and blue, respectively).

Diagram 5: Reflex port close-mic measurements, with the Unit-4 playing at 80dB SPL at 1m (red) and 90dB SPL at 1m (blue).

is significant low-frequency compression when the Unit-4 is working hard. The cause of the compression is likely to be partly down to the reflex port losing linearity, but I'd also expect the Unit-4 to include some compression engineered into its signal processing in order to protect the bass driver from over-excursion. The Unit-4 simply doesn't have the diaphragm area, excursion capability or amplifier power necessary to generate 90dB SPL at, say, 50Hz, so something has to give. And the 105dB maximum level claimed on the Unit-4 spec? AIAIAI don't specify the measurement distance or frequency band for a good reason, because it's pretty unlikely, I think, that a Unit-4 will manage 105dB at 1m. Lastly on Diagram 5, the port tuning frequency is revealed to be around 50Hz, and that means the Unit-4 is unlikely to be able to generate much output below that frequency.

Listening In

The measured performance of the Unit-4 suggest a well designed and engineered, but very small, monitor. And in use, that's pretty much what it sounds like. At modest volume levels, for very nearfield listening, the Unit-4 performs admirably and reveals itself to be a genuinely capable and engaging proposition. It fundamentally reaches the kind of performance level that would make reliable composition and mix decisions perfectly feasible. And that's quite an achievement, and perhaps the most important thing. The Unit-4 doesn't quite, to my ears, have the detail-resolving power of serious high-end nearfield monitoring, and it has a hint of boxy subjective character in the midrange, but its bass performance, at least at modest volume levels, is remarkable given its size (although doesn't sound quite as extended as the 40Hz spec might suggest). The Unit-4 tonal balance is slightly bright to my ears, especially with the grille fitted, but its app EQ can do an effective adjustment job on that. If there's a down side to the Unit-4 it's that it can audibly run out of port linearity, amp power and bass driver excursion, so very high volume levels aren't really in its comfort zone; I feel it begins to sound compressed and choked when pushed, perhaps as some driver protection begins to kick in. Having said that, it's not really reasonable to expect such a small monitor to do high volumes without some evidence that it's working hard. The laws of physics are immutable, but if I were travelling and needed a simple, reliable and portable monitoring solution, I'd be very, very happy with a pair of Unit-4s.

Finally, before I wrap up with a conclusion, the Unit-4 wireless functionality works well and is easy to set up, but I did find myself wondering just how relevant or useful wireless monitoring is in a recording and mixing context. Maybe I'm just too traditional and tied to the cable paradigm, but certainly if I were writing on the road rather than just mixing in the box and had a DAW interface to consider along with my Unit-4 monitors, I suspect I'd most likely go with the setup simplicity of wired connections. I just don't quite see what the benefits of wireless monitoring would bring, other than the need for fewer cables. Your mileage may vary however.

The Unit-4 is a very competently designed and engineered product that does what it sets out to do extremely well. It's limited in some respects, but the limitations are fundamentally a result of its small size, modest amp power and driver dimensions — and it wouldn't be so portable (or cute) if it were bigger. Come to the Unit-4 expecting full-size monitor performance and maybe you'll be a bit underwhelmed. But come to the Unit-4 needing a really compact, portable, and, it has to be said, achingly cool-looking pair of monitors for a tiny work space or on the road, and I think you'll have struck gold. As I said, Denmark is on a roll.

$ $800 per pair.
W aiaiai.audio

Oberheim TEO-5

Polyphonic Synthesizer

Far from being a cut-down version of one of its bigger siblings, the TEO-5 is a brand new Oberheim synth with a personality of its own.

GORDON REID

Back in the not so dim and distant past, the engineers at Sequential took one of the company's synthesizers — the Prophet 6, to be precise — and whipped out all of its voice boards. But, realising that a mute synth is only slightly better than a deaf sound engineer, they then inserted a bunch of different voice boards based upon Oberheim's legendary SEM module, made a few other changes, and revealed a new instrument that they called the Oberheim OB-6. I reviewed this in 2016, and was impressed. I had liked the Prophet 6, but I liked this more.

Let's now fast-forward to the present, where you'll find me noodling around with a new Oberheim synth called the TEO-5. At the time of writing, it hasn't been announced, but a quick glance suggests that Sequential have pulled the same trick here as they did eight years ago. Just look at the chassis and control panel; the TEO-5 seems to have the same relationship with the Take 5 that the OB-6 had with the Prophet 6. But appearances can be deceptive. While there are similarities, even to the extent that the draft manual contains accidental references to the earlier model, don't be fooled into thinking that they are different incarnations of the same underlying five-voice polysynth. Sequential

have no doubt saved time and money by re-employing some existing tooling, but the two are distinctly different instruments.

The Voicing

Like the Take 5, the TEO-5 uses a pair of SSI2130 analogue oscillators for each voice, but that's where the similarities in the audio path end. Whereas the earlier synth used SSI2140 filter chips, the TEO-5 offers a version of the state-variable 12dB/octave SEM filter per voice. Furthermore, there's nothing on its motherboard that looks like a VCA stage, so I contacted the chaps at Sequential, and they confirmed a growing suspicion: unlike the Take 5, there are no audio signal VCAs; loudness shaping is performed in the digital realm. This was not what I had expected and it means that, unlike some other polysynths that I've reviewed in recent years, removing the digital effects from the TEO-5's signal path doesn't give you an analogue synth.

Both oscillators in a voice offer three waveforms — triangle, ramp and variable-width pulse — and you can select any combination of these that you wish. PWM is obtained independently for each oscillator from the modulation matrix (see box) and you can select one or more simultaneous sources to control this. Osc 1 also features a square-wave sub-oscillator one octave below the selected pitch. You can tune each oscillator over a range exceeding five octaves, although I found that — perhaps unsurprisingly — MIDI note C5 wasn't exactly five octaves above C0 on any of the voices in the review unit, and remained a few cents out no matter how many times I calibrated the synth. You can also detune Osc 2 for the usual rich, chorused effects. As you would expect, the cross-mod lacks the consistency of a digital FM synth, but you can use cross-mod and sync simultaneously to create sounds that you might not expect from an instrument of this nature. You can disconnect each oscillator from the keyboard and MIDI to use as a drone or, in the case of Osc 2, as a static audio-frequency modulation source. You can also set the portamento time independently for each. In addition, you can send the output from Osc 2 directly to the amplifier, bypassing the filter. This is an old CS-80 trick that allows you to reinforce the fundamental after heavy filtering, but is more advanced in the TEO-5.

You'll find the source mixer in the filter section. This only offers on/off buttons for the oscillators, the sub-oscillator and your choice of pink or white noise, but you can set the levels for each in the Program menu. There's no external audio input. The SEM-inspired filter itself has bags of character. This isn't surprising, because a quick inspection suggests that it's based on the OB-X-inspired filter in the OB-X8. However, like its forebears, you can't force it to oscillate, even if you use multiple modulators to increase the amount of resonance. (I tried.) If offers four filter types, smoothly transitioning from low-pass to notch to high-pass, with an additional band-pass mode available from the Program menu. You can modulate the filter type, cutoff frequency and resonance using the matrix.

The rest of the signal path for each voice comprises an A-D converter followed by four digital stages. The first of these is the digital amplifier that performs the role of the audio VCA in a pure analogue synth. Following this, the signal reaches the first of three digital effect units. This is a simple overdrive that — far from generating the unholy rasp obtained from some earlier generations of digital distortion — can add a pleasing body and drive to your sounds. This is followed by an Effect unit (great name!), which allows you to select one of 12 algorithms — four types of delay, chorus, flanging, an emulation of the classic Oberheim phaser, a high-pass filter, distortion, an emulation of the original Oberheim ring modulator, rotary speaker and 'lo-fi', the last of which combines overdrive and wow/flutter to emulate a sickly tape machine. There are only three parameters to control each of these, and you adjust them using knobs called Time, Depth/Mix and FBack/Misc. For the most part, these names are descriptive of their functions but, were I ever to use the TEO-5 in anger, I would print the parameter chart, laminate it and stick it next to the synth. Finally, there's a dedicated reverb with controls for the mix, room size, pre-delay and decay time, and a multi-mode HP/LP filter to shape the reverberated signal. Having reached the end of the signal path, the audio is then converted back into the analogue realm before being presented to the audio outputs on the rear panel.

Modulation & Shaping

The primary modulators in the TEO-5 comprise the global LFO 1 and the per-voice LFO 2. These generate the same range of five waveforms as each other, with variable slew to modify the wave shapes, plus controls for rate (reaching as high as 500Hz) and depth. Despite their seeming simplicity, their flexibility becomes apparent when you start to assign them in the matrix. You can even direct their outputs to their »

Oberheim TEO-5
$1499

PROS

- It's a genuine Oberheim at a fraction of the cost of the OB-X8 or OB-6.
- It sounds excellent — in fact, it's hard to obtain a naff sound from it.
- It's far more powerful and flexible than is apparent.
- It feels solid and robust throughout.
- Despite being compact, it has an internal power supply (yippee!).
- Some people will find the narrow design convenient and a strong reason to consider buying it...

CONS

- ... although others will find the keyboard to be too narrow.
- Some players will dislike the positions of the performance wheels.
- Five voices will be too few for some uses.

SUMMARY

The TEO-5 is a lovely synth to program and, when you've done so, it sounds great. There's much to recommend it and very little to criticise. Although its physical design will not be to everyone's taste, it deserves to be a significant success.

» own inputs for all manner of uncommon modulation effects.

You can also create cyclic waveforms using the contour generators, of which there are two per voice. Used conventionally, these generate five-stage (DADSR) bipolar contours with, by default, Env 1 directed to the filter cutoff frequency and Env 2 to the audio amplifier. Nonetheless there are two other routing options: Env 1 acting as an auxiliary contour generator assigned within the matrix and Env 2 controlling both the filter cutoff frequency and the amplifier gain; and Env 1 acting as an auxiliary contour generator and Env 2 controlling just the filter cutoff frequency while the audio amplifier is gated open when you press a key. (It's harder to describe than it is to use.) Given that the contours are always available as matrix sources and can be assigned multiple times, the system is considerably more flexible than the control panel might suggest. Oh yes, and returning to the concept of using the contours as cyclic oscillators, you can program Env 1, Env 2 or both to repeat indefinitely and, with short attacks and decays, to generate audio-frequency waves that you can use for things such as AM and FM synthesis. Both contour generators are velocity-sensitive, and you can direct velocity to their amounts on the front panel, and to their times and sustain levels using the matrix. This is particularly useful for things such as the synthesis of plucked instruments.

More Goodies

Inevitably, the TEO-5 offers much more than an audio signal path with a couple of digital LFOs and contour generators stuck on the side. For example, there's a monophonic unison mode plus an associated chord mode invoked by holding a chord while pressing the Unison button. Since the TEO-5 is responding monophonically in both these cases, it's good to see that it provides low-, high- and last-note priority with either single- or multi-triggering so that you can ask it to respond like your favourite monosynth. Then there's the Vintage knob, which adds variation to the pitch, filter cutoff frequency and contour times on a voice-by-voice basis. Used gently, this can impart the usual, subtle inconsistencies that add interest to a sound. Interestingly, Vintage can be a modulation destination, so I created a simple lead patch, selected five-voice unison with minimal detune, and then controlled the Vintage amount using the mod wheel. The results were great, allowing me to take a well-behaved sound into new, thicker realms to accentuate the climax of a solo before pulling it back into line at the end. That's worth some experimentation.

Low Split is another interesting facility. This allows you to select a split point on the TEO-5's keyboard but, rather than letting you place different sounds on either side (you can't... the TEO-5 is strictly monotimbral) it allows you to transpose the lower side of the split down by either one or two octaves. This increases the effective width of the keyboard to 4.5 or even 5.5 octaves, albeit with a one-octave or two-octave hole in the middle. If your music involves playing a bass note with your left hand and a triad or two with your right, I can see this being very useful. Happily, the correct MIDI note numbers are sent in both cases.

Talking of tuning, the TEO-5 offers 64 memories for alternative scales, and you can save your selection from these within each Program. The choices range from subtle variations on conventional Western tuning, to Eastern scales, to some wildly experimental options, and the manual explains each of these in detail. It's well worth experimenting with these even if you don't expect your next Top 10 hit to exploit the subtleties of Helmholtz's untempered 24-notes-per-octave Harmonium Tuning.

In Use

When you look at the TEO-5 from a player's perspective, it's apparent that it's a new format for an Oberheim polysynth. All previous models sported either four-octave »

The Modulation Matrix

Although the TEO-5 appears to be a simple synth, its 19-slot modulation matrix elevates it into complex sound-design territory. The first 16 of these slots are freely assignable, drawing from a list of 19 sources (some global, some per-voice) and 64 destinations. The other three slots are dedicated to the global LFO 1, the polyphonic LFO 2, and the auxiliary envelope. You can often select a source by holding the Source button and adjusting an appropriate control, and select a destination in the same fashion using the Destination button. Nonetheless, there are some options that you can access only through the menus. Interestingly, the amounts of each of the first 16 slots can be the destinations of the other slots, which means that you can create some unexpectedly convoluted modulation routings. What's more, you can program multiple sources to affect a given destination and simultaneously direct a single source to multiple destinations. If you make full use of velocity, aftertouch, the wheels and the pedal inputs, the TEO-5 proves to be a remarkably complex and expressive instrument despite its seeming simplicity.

The Rear Panel

The TEO-5 offers a simple rear panel. Starting on the left, you'll find the digital I/O: USB for connection to your computer (no drivers needed for either Mac or PC), plus a full set of 5-pin MIDI In, Out and Thru. Next come the two analogue control inputs: Footswitch, which is typically used for a sustain pedal but is also able to latch the arpeggiator and start/stop the sequencer, and an input for an assignable TRS expression pedal. Finally, there are the holes through which you hear the sound: unbalanced quarter-inch left/right audio outputs plus quarter-inch stereo headphones. (As always, I would have preferred the headphone output to be at the front of the synth.) More good news is the IEC mains socket that feeds the internal, universal power supply. There must have been a great temptation to use an external PSU on a synth of this size, and it's to the designers' great credit that they didn't succumb to this.

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The TEO-5's front panel. Did you know the '80s business synth look is making a comeback?

» keyboards (the 4-Voice, 8-Voice, OB-SX, OB-6 and — if you'll allow it — the OB-12) or five-octave keyboards (the OB-X, OB-Xa, OB-8, Matrix 6 and 12, and the OB-8X). Only the mono- and duophonic synths (the OB-1 and 2-Voice) were narrower, and all 13 instruments had their performance controls to the left of the keyboard itself. Yet here we have a polysynth no wider than some monosynths, with its pitch-bend and modulation wheels placed behind the keyboard. Why? I suspect that it's because, when designing the Take 5 chassis, Sequential found that they could build (and therefore sell) a synth of this format more cheaply, broadening its appeal as an affordable option. This probably explains why the TEO-5 also has plastic cheeks rather than wooden ones. Nonetheless, it feels very robust, and there's not a wobbly knob to be found anywhere on it.

Happily, programming the TEO-5 is easier than its wealth of functions might imply. Despite its diminutive size, its panel is clear and accessible with most of the controls being of the one-function-per-knob or per-button variety. Nonetheless, there are important facilities that can only be accessed via the matrix and the two menus. (The Program menu offers 33 parameters, some of which are already present on the panel, while the Global menu offers a further 34 that allow you to configure the TEO-5 and tailor things such as the velocity and aftertouch responses to your taste.) Sure, the OLED is tiny, but this screen size has become the norm for many synths and rackmount modules in recent years and, as long as I had my reading glasses to hand, I found it to be clear and simple to use.

So now it was time to program and listen and, of course, ask the crucial question: "Does it sound like an Oberheim, or is it a generic polysynth painted with blue pinstripes in the hope that you'll listen with your eyes?". Rather than step through the factory sounds (that came later), I spent many hours creating sounds that might otherwise cause me to turn to one of my vintage Oberheims. I soon discovered that, while the filter in the TEO-5 places it firmly in Oberheim territory, it's not a miniature version of an OB-Xa or OB-8; nor is it a low-cost version of the more recent OB-6 or OB-X8. In fact, the more I used it, the more I realised that it's a new and enhanced Oberheim, not a slavish

Arpeggiators, Sequencers & Rhythms

The TEO-5's monophonic arpeggiator offers five modes, a maximum of three octaves' range, up to three repeats per note, and Hold. I was pleased to find that the resulting arpeggios can be transmitted over MIDI so that you can drive other equipment in real time, or save interesting arpeggios on your DAW and direct them elsewhere if you decide to use a different synth in the future. The 64-step sequencer is equally simple. It allows you to store up to five notes per step, and you can program ties and rests, but that's all. There are no editing capabilities, so the only things that you can change are the playback tempo and the transposition. Happily, sequences are 'per program', so you can include one with every sound that you create. Like the arpeggiator, the sequencer can send its output to MIDI. If you have no hand available, you can control both the arpeggiator and the sequencer using a footpedal, which is always a good thing. The frequency of the internal clock that drives these ranges from 30bpm to 250bpm, with a wide range of clock divide ratios. You can also tap the cunningly named Tap Tempo button to determine the clock frequency, or replace it altogether with received MIDI Clock. As you might expect, the clock can also be used to synchronise the LFOs and delay effects, and you can invoke different ratios at each of its destinations to generate interesting rhythmic effects.

homage to the past, and that it has a strong character of its own. I programmed classic polybrass patches, synth strings, slow pads, chimes, percussion, effects, arpeggios, mini sequences and more and, while the TEO-5 isn't going to be your go-to keyboard for things such as e-pianos and organs, it proved to be remarkably flexible and it almost always sounded great.

It also turned out to be a fabulous monosynth. I created a simple two-oscillator patch with two-voice unison, maximum filter resonance and the filter cutoff frequency controlled by both contour generators to create a complex 'blip' and sweep. I then directed aftertouch to the LFO 1 amount as well as to the filter cutoff frequency and audio level, and added a slow chorus plus a touch of reverb. Whether used for bass duties or tweaked to grace a proggy magnum opus, the results were superb. To be honest, I could have widdled all week with huge unison leads, sync leads, monster basses and more had I not had to take my hands off the keyboard to type this review.

Of course, it's fair to ask why Sequential implemented such an unusual signal path in the TEO-5. There were probably several reasons, although the most obvious is that it's cheaper to build a polysynth with digital LFOs, contour generators and amplifiers than with analogue ones. Another argument goes something like this... Depending upon various aspects of the design, the differences between an analogue oscillator and a modelled one can have an audible impact on some sounds. Likewise, the differences between an analogue filter and a modelled one. In contrast, the impact caused by replacing a VCA with a digital amplifier, and replacing analogue contours with digitally generated ones, may be far less apparent. Whether this is true depends on the nature of the design. If the digital amplifier is programmed to be linear and free of all noise and distortion it would, perhaps, replace the perceived 'warmth' imparted by an analogue VCA with digital 'transparency'. But if the system is modelled to replicate the response of a given VCA and a specific analogue contour generator, replacing one technology with the other may make little difference. My experience with the TEO-5 is that it's a complete non-issue; although I knew otherwise, nothing in its sound ever suggested to me that the primary audio path was anything other than analogue. Actually, that's not quite true. The TEO-5 is deliciously devoid of background noise and hiss. If that's a consequence of the digital stages, sign me up as a fan!

Finally, I had to conduct a few functional tests, just to make sure that everything was doing what it should. I couldn't check everything in the time available, but all seemed well and, while testing the MIDI and USB connectivity, I realised something rather excellent: there was no trace of the dreaded USB ground-loop hum. That was the icing on the cake!

Conclusions

Before I played it, I must admit that I wasn't particularly turned on by the TEO-5, but that was a consequence of its narrow keyboard, the positions of its wheels, and the lack of either poly aftertouch or MPE. That mood evaporated as soon as I discovered how easy it is to program and how classy it sounds. Sure, if I were to use one as a polysynth, I would play it from a 76-note controller and enjoy its sounds on a grander scale, but if your space is constrained, its size could be the attribute that allows you to use a genuine Oberheim where it may not otherwise be possible. There's a great deal to like about the TEO-5 and, now that it's time to hand it back, I'm going to have to admit that I'm rather turned on by it.

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Shure Nexadyne

Dual-capsule Dynamic Microphones

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PROS

- Crisp, clear, detailed sound.
- Unusually consistent polar patterns with excellent rejection at the nulls.
- High-class build quality.
- Controlled proximity effect.

CONS

- The built-in presence boost might not work equally well for all singers.

SUMMARY

Shure continue to push the boundaries of moving-coil mic design, and their efforts continue to bear fruit! The Nexadynes are dynamic mics with directional properties you'd normally find only in a capacitor mic.

SAM INGLIS

Over the years, Shure have produced microphones using all of the mainstream operating principles. Their current range includes the highly regarded KSM353 and 313 ribbon designs, as well as numerous large- and small-diaphragm capacitor models. But if there's a particular technology with which the Shure name will always be associated, it's the moving-coil dynamic mic. Many of the most important developments in dynamic mic design originated with the company, perhaps most notably the Unidyne cartridge — the first moving-coil capsule to have a native cardioid response. All of the hundreds of millions of directional dynamic mics in daily use today owe their existence to this 1930s breakthrough by Shure designer Ben Bauer.

What's more, unlike some contemporary manufacturers of dynamic mics, Shure are continuing to innovate. They don't see the moving-coil microphone as a mature technology to be endlessly repackaged, but as a field where development is still ongoing, and one where cutting-edge research into electro-acoustics and materials science can still yield measureable and audible benefits.

This research bore rich fruit in 2016 with the KSM8, a stage vocal microphone that debuted Shure's novel Dualdyne capsule. As the name suggests, the Dualdyne capsule extended into the moving-coil realm a concept that has long been standard in the design of large-diaphragm capacitor mics: the addition of a second diaphragm at the rear of the capsule to help control directionality and proximity effect. As Hugh Robjohns explained in his review of the KSM8 (www.soundonsound.com/reviews/shure-ksm8-dualdyne) this was an extremely impressive feat of engineering on Shure's part, and one that required seven years of research.

Doubling Up

Eight years on, we are now seeing the next stage in this development. The Nexadyne range employs another breakthrough Shure capsule design, which the company are dubbing Revonic. The rear diaphragm in the Dualdyne capsule is passive, in the sense that it modifies the acoustic properties of the system but does not itself act as

a transducer. By contrast, the Nexadyne assembly actually includes two entire capsules: a conventional one at the front of the mic and a second one behind it.

With the standard Braunmühl-Weber capsule used in most large-diaphragm capacitor mics, it makes sense to think of the front and rear as being more or less independent cardioid elements, and indeed, variable-pattern small-diaphragm mics such as the Neumann KM88 and KM86 actually feature two completely self-contained cardioid capsules mounted back-to-back. The directional qualities of the mic as a whole are modified by combining these in different amounts: for example, summing them equally delivers an omnidirectional response, whilst summing them equally in opposing polarity gives figure-8 pickup.

The two capsules in the Dualdyne assembly are summed in opposite polarities, but it would be misleading to think of them in the same way. For a start, both of them face forwards, and they're not designed to be operated or addressed independently. Instead, they form a complete system that has a fixed polar pattern. Within this system, and in comparison to a conventional single-element mic, the second capsule has several important functions. It plays a very significant part in defining said polar pattern, and especially in making this consistent throughout the frequency spectrum. Connecting it in reverse polarity eliminates the need for the humbucking coil found in single-element mics, since any interference is picked up equally by both capsules' voice coils and cancelled out. The same feature also helps to minimise handling noise: a knock or bump on the mic will set both capsules moving in very similar fashion, so the net output of the two capsules summed in reverse polarity should be quite low.

As with the Dualdyne, the Nexadyne engine is a supremely high-tech affair. The internal acoustics are so complex that it would have been completely impossible to develop something like this without advanced computer modelling and simulation. It demands incredibly precise manufacturing to extremely fine tolerances, with individual parts such as diaphragms barcoded and traced through the production line, in order to be matched as accurately as possible.

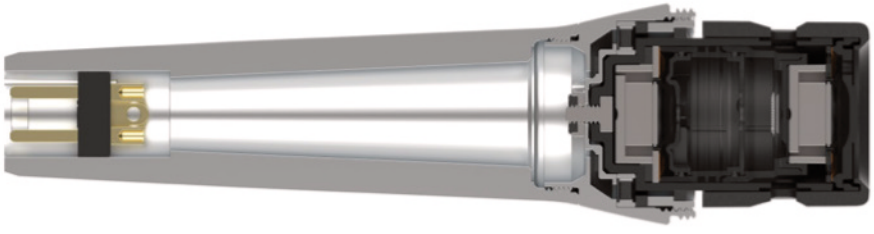
The supercardioid Nexadyne 8/S and the cardioid 8/C are housed in the same shell, with the main visual difference being the extra ribs along the grille in the former.

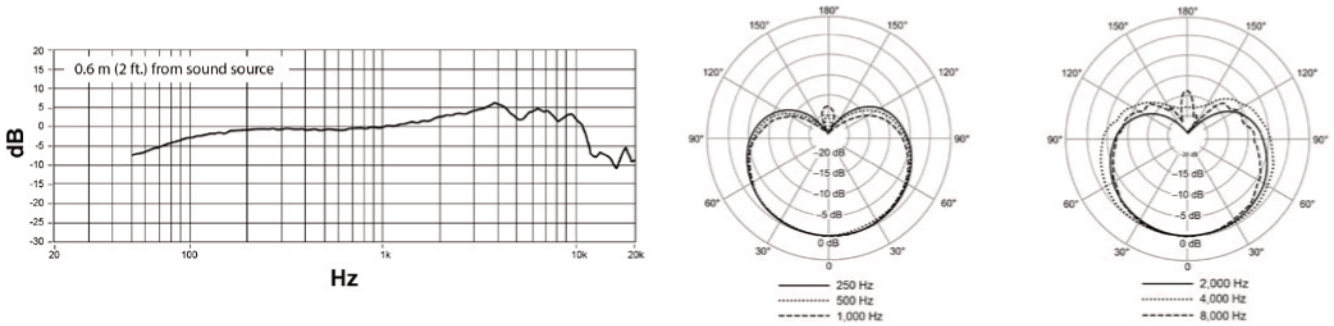
Nex Level

At launch, the Nexadyne range comprises cardioid and supercardioid models called, respectively, the Nexadyne 8/C and Nexadyne 8/S. These are available as wired mics with conventional XLR outputs, or as heads for wireless systems. The former were supplied for review. They come in compact, robust zip-up cases, with a simple mic clip included. Commendably, the outer packaging has been designed with sustainability firmly in mind, and consists only of a simple printed card box.

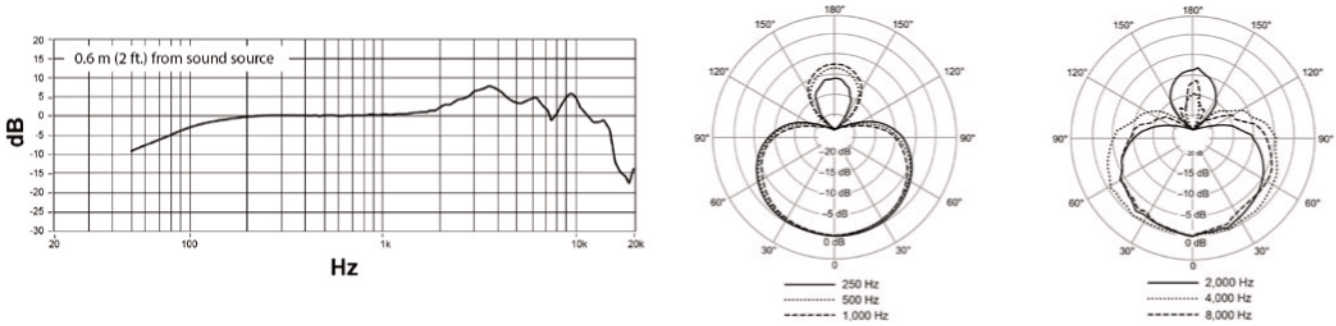
You'd expect a new stage mic from Shure to be bulletproof in construction, and the Nexadyne models certainly appear to be. The die-cast aluminium shells are finished in a very smart and hard-wearing black, which is the only option for the wired mics; the wireless heads are also available in a nickel finish. Both mics adhere to the now-standard 'ice cream cone' form factor, and coincidentally, are almost identical in size and shape to the Earthworks SR117 reviewed by Paul White in May's *SOS*. That means they are slightly on the large side as stage mics go, measuring around 200mm long by almost 50mm in diameter at their widest point. The supercardioid version is slightly the heavier of the two, at 294g to the cardioid's 258g, but both sit very comfortably in the hand. Apart from the small gold logo near the base, the main visual identifier that distinguishes the two versions is the number of ribs that run along the side of the grille. The 8/S has four of these to the 8/C's two.

The reason for the Nexadynes' ample girth becomes apparent when you unscrew the tightly woven black grille. The shockmounted capsule assembly is considerably larger than those of most dynamic mics, in every dimension. The assembly in the supercardioid version is several millimetres longer than that of the cardioid; both have four rectangular, mesh-covered 'windows' along the sides, and these again are larger in the supercardioid mic. Happily, there's no


The dual forward-facing capsules are clearly visible in this cutaway view of the cardioid Nexadyne 8/C.



Frequency response and polar pattern charts for the cardioid Nexadyne 8/C (above) and supercardioid 8/S.



» foam to decay and cause problems down the line: the headbasket is lined instead with a flexible gauze-like material.

In some ways, the Nexadyne specs are what you'd expect of a handheld dynamic mic. Sensitivity is quoted as 2mV/Pa for the cardioid and 2.8 for the supercardioid, and the on-axis frequency response is familiar, being relatively flat through the low mids before a broad presence peak sets in. In the cardioid version, this reaches a 6dB climax at about 4kHz, with secondary peaks at about 6.5 and 9 kHz; in the supercardioid model, the primary peak is almost 8dB up at 3.5kHz. The high-frequency roll-off sets in at about 10kHz on the cardioid and 12 on the supercardioid, though both have usable response up to 20kHz.

Other specifications, however, are less typical. Both models are transformerless and have a higher than usual output impedance, with the 8/C quoted as 300Ω and the 8/S as 450Ω. But the most eye-catching feature is the polar pattern graph. This is unusually consistent across the frequency spectrum, especially for the cardioid model, where the lines stay within a 5dB range for practically all angles of incidence except directly behind the mic. It's a shame that Shure haven't included a 125Hz plot, though, as it would be interesting to see what happens to the pattern at low frequencies.

On Point

I've never tried the KSM8, but by all accounts it's notable for a relatively neutral sound without an obvious presence peak; some have even called it 'condenser-like'. Shure are clearly going for something different with the Nexadynes, however. That presence peak is readily apparent, and the on-axis sound is crisp and bright. It's what you might call a 'mix ready' sound, and unlikely to require much additional treble boost at the console. Indeed, the supercardioid version was actually a touch too lively on my voice, which usually welcomes plenty of help in the upper midrange.

"It's what you might call a 'mix ready' sound, and unlikely to require much additional treble boost at the console."

It's a familiar tonality, and if you stayed perfectly still at an optimum distance from the mic, and there was nothing else going on in the room, there perhaps wouldn't be an obvious difference between the Nexadyne and many generic dynamic stage vocal mics. But as soon as you start to move around, you begin to appreciate the benefits of the Revonic dual-capsule design. It doesn't eliminate proximity effect, especially on the supercardioid version, but you can work the mic right up close without things becoming at all muddy, and you can pull back from it without losing all the body from the vocal sound. The cardioid version is also very tolerant of side-to-side movement, in a way that many stage mics aren't. Whether you're a singer who deliberately employs 'mic technique' or someone who simply moves around a lot on stage, I think

As this exploded view of the 8/S suggests, the Nexadyne mics are highly complex and require cutting-edge tooling and assembly.

the Nexadynes will prove a lot more forgiving than conventional vocal mics.

As a point of comparison, I dug out my Shure Beta 57A, and the comparisons were enlightening. Both Nexadynes are a fair bit brighter than the 57A, and whereas the older mic can get muddy in the 200Hz area when used close, the newer ones stayed crystal clear. What was really eye-opening, though, was the way the mics capture sound off-axis. In many cases, descriptions like “cardioid” and “supercardioid” should really be paired with qualifiers such as “at some frequencies, sometimes”. Not so with the Nexadynes. The cardioid model exhibits impressive rejection at the rear, with a large and even pickup zone at the front, just as it should, while the 8/S likewise comes much closer to the theoretical ideal than the nominally supercardioid Beta 57A. At the angles of greatest rejection — 180 degrees for the 8/C and 126 degrees for the 8/S — the nulls are impressively deep.

The flip side of this is that, in relative terms, the presence peak around 3-4 kHz is even more pronounced off-axis, and such sound as is captured at the nulls isn’t terribly attractive. Nevertheless, I think the predictable and consistent polar pattern, strong rejection in the nulls and minimal proximity effect are absolutely a winning combination. Handling noise isn’t easy to measure objectively, and subjectively I didn’t perceive it as being lower with the Nexadynes, but I think that mainly reflects the fact that the Beta 57A is already very good in this department. The grille and gauze combo does an effective job of minimising plosive artefacts, though a singer who works the mic very close will still generate some pops.

The Keys Test

I don’t do live sound all that often myself, so to give the Nexadyne a proper workout in a real-world situation, I lent the cardioid version to a friend who runs a hard-working party band. One of the biggest challenges they face is finding a vocal mic that works for a singing keyboard player. When your hands are flying all over the keyboard as you crank out ‘Crocodile Rock’, it’s almost impossible to address the vocal mic in a consistent, close-up way. The verdict on the Nexadyne 8/C? “Absolutely incredible. It’s a real cracker in a live situation. Very little EQ required, and had a really great capture area for the keys player.”

Price-wise, the Nexadyne mics clock in at around twice as much as the Beta 57A and 58A, around the same as Shure’s Beta 87A capacitor vocal mic, and somewhat less than the aforementioned KSM8. If you’re choosing for your own voice, then obviously there’s no substitute for trying them out: the Nexadynes’ pronounced treble boost will work better for some singers than others, who might suit the KSM instead. But if you’re a live sound engineer or venue buying mics that will be put up on stage every day to deal with all-comers, the Nexadynes have the potential to make your life a whole lot easier.

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EastWest Hollywood Strings 2

Sample Library

EastWest get intimate with their latest orchestral string section library.

JOHN WALDEN

EastWest's Hollywood Orchestra Opus Edition — reviewed by Dave Stewart in the July 2021 issue of *SOS* — is undeniably epic. With approximately 1TB of sample data, it's epic in size. Thanks to the super-slick Opus front end, it has a pretty epic user interface. And, developed in collaboration with expertise from Sonuscore, its Orchestrator performance engine can generate epic orchestral arrangements from even simple MIDI chord sequences. Oh, and don't forget the sound; yup, that's definitely epic also. This is a premium orchestral sample library product capable of seriously good results.

All of which begs an obvious question; what can you do to better 'epic'? Well, when it comes to the orchestral string section at least, we are about to discover, as EastWest have just released Hollywood Strings 2. So, let's raise the baton and see what they have done to raise the bar.

Hollywood Strings 2 delivers a close-up and intimate sound thanks, in part, to the additional close mic options that can be seen in the Microphones mixer panel.

Size Matters

As befits the 'Hollywood' title, the original Hollywood Strings library had a very cinematic sound. Indeed, it's apt for the whole Hollywood Orchestra; if you want that blockbuster film score sound from your virtual orchestra, EastWest's flagship library can deliver it.

With that box already ticked, rather interestingly, EastWest decided to take a somewhat different direction with Hollywood Strings 2. The product's tagline — 'close-up and intimate' — reflects this and the library is based upon a fairly compact 21-piece chamber strings ensemble consisting of six first violins, four second violins, four violas, four celli and three basses. The instruments were recorded in EastWest's own Studio 2 which, while used for soundtrack work, is also used for bands (Green Day, Metallica and the Red Hot Chili Peppers have all recorded there, for example) and provides a more intimate sound than a larger recording space would.

The 'intimate' isn't just about the space, though. While three arrays of mics provide a room sound — main, midfield and surround — every instrument was also close-miked, including mics attached directly to the instrument. Within the Opus mixing controls, these provide close main, close secondary and close overhead options alongside the room/surround mics. If you want a tighter, drier, more intimate sound, Hollywood Strings 2 should therefore have the underlying sample base to provide that while still letting you 'go big' with the room mics and suitable reverb when required.

And, talking of size, Hollywood Strings comes in at just over 160GB so, from a technical perspective at least, perhaps not quite so small and intimate. That said, realism and detail in your virtual instruments generally increases with a larger sample base, all other things being equal.

Ensemble Options

As with the original HS1, the bulk of the presets for Hollywood Strings 2 are divided into instrument-specific ensembles. Within the Opus browser, for each of these instrument ensembles — first violins, second violins, violas, celli and basses — their respective presets are subdivided into five performance style groupings; long, short, effects, legato and keyswitch. In turn, each of these preset groups then provide individual articulation presets, while the keyswitch groups offer a single preset with multiple keyswitchable articulations. Composers often have their own preference for working with either single articulation or multi-articulation presets, so it's good to have that choice straight out the box.

For those occasions when a full string ensemble patch is required, a further group of Full Strings presets are provided and divided into long, short and effects

groupings. Each of these offers a range of individual articulation presets and, when loaded, open as multiple instruments within the Opus' Instrument Rack providing violins, violas, celli and basses. Each is configured to use the same MIDI channel and, as you play, notes are allocated to each sub-ensemble based on pitch. While most of these are single-articulation presets, there is a very useful Full STR Stac Sus preset that offers both staccato (with three dynamic layers) and sustain (with four dynamic layers) articulations with automatic switching based upon how you play. Either velocity (for staccato) or CC11 (for sustains) are used to control dynamics, and for the sustained notes, you can add vibrato via the mod wheel. It's a very playable preset and great for sketching out initial ideas. For the ultimate convenience on that front, there is also the Live 21 Piece Ensemble preset, which loads as a single instrument. It's built on just sustains but, with a pretty rapid attack, you can easily fake shorter notes, whether that's for actual live performance or for sketching ideas.

As with HS1, once you load an HS2 instrument into Opus, you can switch that preset between Soft, Classic and Epic moods. This adjusts the combination of microphone mix and reverb (and some other settings under the hood) to quickly change the character of the sound. In HS2, the Soft mood favours the close and mid mic positions, Classic combines close and main, while Epic utilises the mid and surround mics, giving a progressively more ambient sound. These three moods provide some very contrasting options, and you can make your own manual adjustments if you need to refine things further.

The Art Of Articulations

As befits a premium string library, HS2 doesn't disappoint when it comes to the array of performance articulations offered. EastWest's website and the HS2 PDF manual provide a great summary of what's included for each of the ensembles, so I'll just provide an example here — the first violins — to give a flavour of the coverage.

The first violins are offered with six different Long articulation options. These includes detache, a stac/sus combo, an expressive sustain, and a preset that uses the mod wheel to move between non-vibrato, vibrato and a molto (wider) vibrato. The Short category offers 10 instruments, with col legno (played with the wooden part of the bow), a selection of marcato, staccato, spiccato (that offer progressively shorter notes) and pizzicato presets. You also get up/down staccato run options to provide super-realistic note transitions in those faster passages.

Eight presets within the Legato category provide different legato variants with bow change, slur (with expressive, molto and repetition options) portamento and trill available. The scripting options allowing your playing style, velocity, CC1 and CC11 to control details of the performance, blending the legato lines with sustained samples and controlling dynamics. There is also an 'All VS' preset that lets you switch between the different legato behaviours based upon MIDI note velocities, while CC11 controls dynamics.

The Effects category provides three presets — tremolo, harmonics and flautando — the latter offering a soft, delicate sound based upon a light bowing technique near to the fingerboard. Finally, within the Keyswitch category, you get a full keyswitchable preset that offers 13 different performance articulations.

In broad terms, the articulation sets across the other ensembles are similar but just some minor differences. For example, the second violins miss the harmonics from the Effects category, while the basses don't offer a trill or portamento legato presets. However, all these other instrument ensembles do offer an additional Bartok pizzicato (a stronger pizzicato where the string often snaps back onto the fingerboard; it's a somewhat more percussive sound) in their respective Shorts category. In summary, HS2 provides an impressive array of playing techniques that ought to satisfy almost any orchestral composer.

EastWest Hollywood Strings 2

$499

PROS

- Distinct sound palette compared to HS1.
- Close miking delivers great attack to the shorter articulations.
- High-quality sounds throughout.

CONS

- Only that it represents a considerable investment as a one-off purchase.

SUMMARY

EastWest's interesting design choices have paid off with Hollywood Strings 2. The more intimate sound, and the excellent close-miking options, deliver something distinct from, but complementary to, HS1. It sounds fantastic.

String Theory In Practice

Of course, what really matters is just how this combination of interesting design choices and technical details combine to create HS2's sound palette. Well, let's start with the obvious; the individual instrument ensembles within Hollywood Strings 2 »

Full keyswitchable presets are also included.

» sound fabulous. The dynamic layers and the preset mood options ensure that every instrument has a real depth to its sound. This enables you to move from the gentlest of musical moods right up to the most epic and bombastic action/drama styles. Deep (and, when required) powerful basses, emotive celli, beautifully balanced violas and violins that can go from delicate to soaring; it's all here for you to exploit in your compositions.

You could (indeed, Dave Stewart in his review did) say the same about HS1, but those close-miking options, the smaller recording space and the more compact 21-instrument ensemble undoubtedly deliver a unique and distinct character in HS2. That's particularly evident in the short articulations that use any of the close-mic options. The note attacks are clearer, seem faster, and offer greater emphasis. Used with a minimum of ambience (from the mics or reverb) and the sound can be very raw and intimate (you get a very clear sensation of the compact size of the ensembles). However, when you blend in the room/surround mics and reverb, providing you keep those close mics within the mix, you can get this very compelling combination of size and immediacy. Comparing a number of the HS1 and HS2 stacatto and spicatto articulations, the faster note attacks in HS2 are very obvious. Faster note runs can often be difficult to pull off in a truly convincing fashion with some virtual string instruments. In these styles, HS2 enables a tighter, more dramatic result; it's very impressive stuff.

Those close-mic note attacks don't just benefit the short articulations though as, when required, they can also add a touch of bite to the start of sustained notes. There are also plenty of cinematic applications for these drier, more immediate, sounds. They simply allow you to create a very different set of styles/moods that perhaps don't fall so easily from the sounds available in HS1. Equally, they open up other creative possibilities such as a drier sound required for use in some more intimate pop contexts, or a more controlled sound if used in a rock/metal mix where you need to place the strings more precisely in what might already be a full arrangement. You might also use just the close mics — which can sound pretty unusual to the ear without any of the room/reverb element we are so used to with orchestral strings — as a DI-style signal into some creative effects chains (maybe an amp sim?) for sound design. For me, this close mic element is an absolute highlight of the HS2 sound.

Full section presets stack individual instruments within Opus' Instrument Rack.

For users of the Orchestrator element within Hollywood Orchestra, it's worth noting that, at present at least, HS2 is not currently integrated into that system. EastWest have not yet revealed their plans on this front but, given just how well received the Orchestrator technology has been, I find it difficult to imagine that support for HS2 will not come. Fingers crossed...

"Deep (and, when required) powerful basses, emotive celli, beautifully balanced violas and violins that can go from delicate to soaring; it's all here for you to exploit in your compositions."

Conclusions

EastWest have made some very astute design choices for Hollywood Strings 2. Yes, it matches (and perhaps exceeds) the high sonic standards set by HS1. However, it's not just a case of 'more of the same, but better'; HS2 brings a distinct sound palette all of its own and, while they have territory in which they overlap, they both have their respective strengths. Users should, therefore, see HS2 as complementary to, rather than a replacement for, HS1.

As a one-off perpetual licence, Hollywood Strings 2 is a pretty significant investment, even if you manage to purchase it at the discounted launch pricing. It is, however, a professional product and priced accordingly. Fortunately, if you are an EastWest ComposerCloud+ subscriber, you will automatically see Hollywood Strings 2 added to your Installation Center library list. I know subscription-based services are not to everyone's taste but, if you are a working composer, the annual subscription to ComposerCloud+ is one heck of a way to manage your budgeting. Given the huge range of top-quality sounds it provides access to, I really do think it has become one of the best deals available.

Hollywood Strings 2 brings something new and unique to EastWest's orchestral instrument line-up and adds a distinct — and very usable — sonic palette to their existing string section options. I appreciate these major libraries are a very significant development undertaking but, having experienced HS2, I'm now eagerly awaiting what might come next for the other orchestral sections; EastWest, bring it on. A worthy addition to the Hollywood Orchestra stable, Hollywood Strings 2 is a bit of a triumph.

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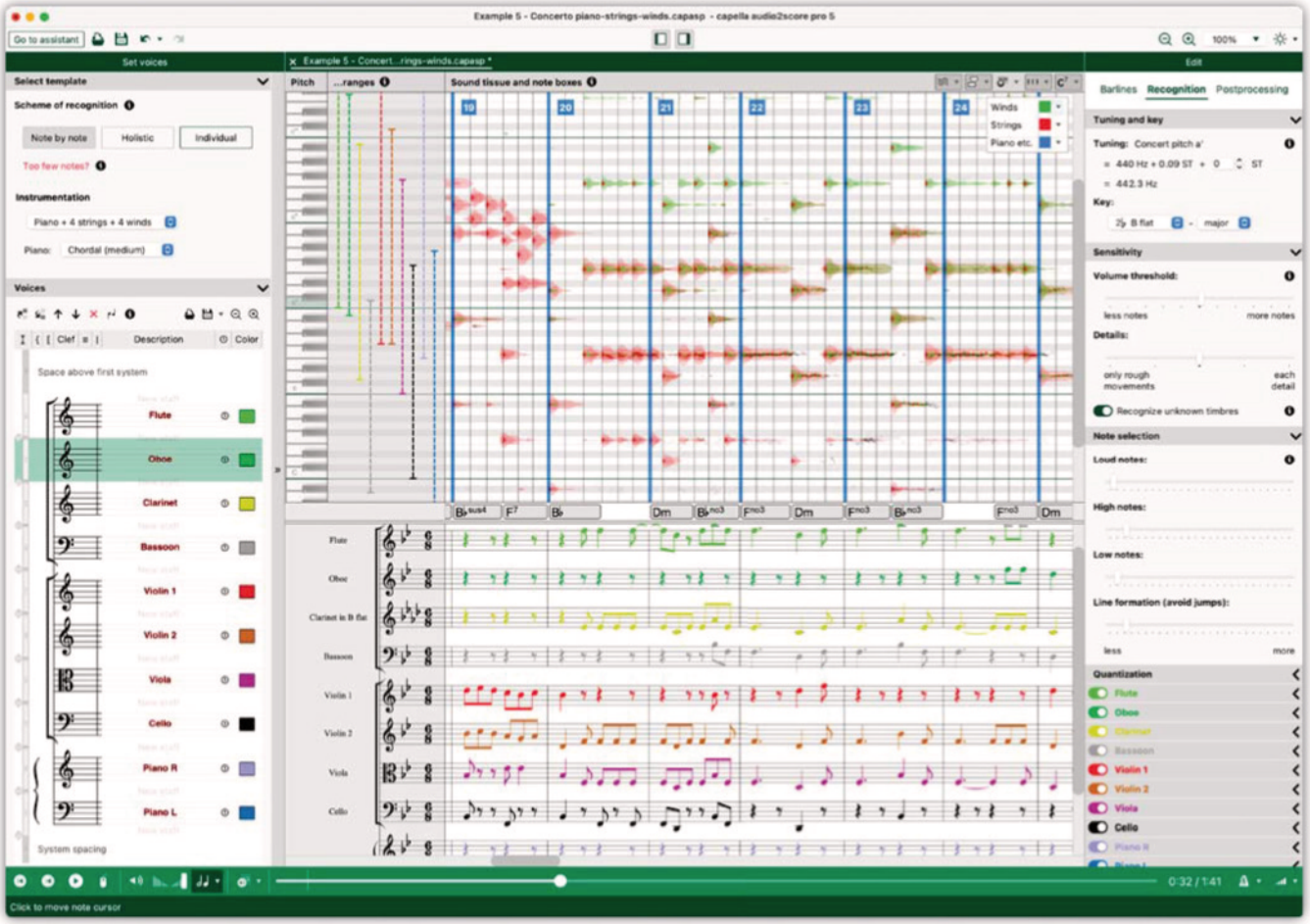
capella audio2score pro 5



Score Generating Software

audio2score uses AI to turn recorded music into sheet music, but just how good are the results?

ROBIN BIGWOOD

While headline-grabbing AI technology fills our social media feeds with fantasy artwork and formulaic poetry and prose, a much more focused version of it has been powering various transcription-oriented music software for years. Tools to scan printed music to live notation, or that extract note information from melodies or individual tracks, ranging from cheap iOS apps to Melodyne, are commonplace now. However, software that can ingest audio of a multi-instrument mix and then attempt to separate out and notate all its constituent parts is still comparatively rare. For good reason: it's really difficult to do, even for expert-level human musicians. But it's exactly this that the (self-consciously lower-case) capella audio2score pro 5.0 aims to do, on Windows (10 or later) and macOS (10.13 or later). In fact the app goes beyond mere transcription, as we'll see in a minute, well into arrangement territory. It's more capable and flexible than you might at first assume.

Hope

audio2score pro 5 (which I'll call A2S for the rest of this review, for grammatical sanity) runs in a single window on your computer, and has a pleasant immediacy about it from the off. After launch you're prompted to open an audio file (an 8- or 16-bit WAV, MP3, or WMA on Windows), or an A2S project in CAPASP format. Then it's straight down to business.

audio2score in full flight, in main program view, showing the detected harmonic content for an orchestral recording in a piano-roll editor, and the resulting live notation below.

Assuming you've fed in new audio you're met with a dialogue box that asks you to specify what it contains: the choices are Piano only, Classical (without vocals), Pop (without vocals), and Pop (with vocals). An analysis process takes place, with a progress window appearing to show the app's progress calculating 'spectrum' and 'sound tissue', then recognising tuning, key and notes. On my M1 Mac this typically took (approximately) half the duration of the audio, varying according to complexity. And then, boom, a notated score appears.

It is just possible, at this stage, especially with simple musical material, that the software will have nailed it. You can gauge that visually if you're a music reader, or

to some extent via A2S's embedded SoundFont-based playback module. Playback controls are found in a strip at the bottom of the window, and one of them is a useful crossfader that outputs your original audio file (in mono) when set to the left, sample playback from the extracted data to the right, or a sync'ed mix of the two in between.

If all is up to snuff you'll have already reached the ultimate point of the exercise, which is to export a PDF score, a type 0 or 1 MIDI file, or a MusicXML document for passing on to another notation application. If you use capella software's own notation application (capella) then the transcription can be transferred to that directly or via the specialised CapXML format. There is one last option, to save the A2S project itself, and that lets you return to unfinished work without having to rerun the analysis process.

Reality

Although audio2score's initial analysis often gets some things right, the chances of complete accuracy are actually slim. And while playback of extracted data frequently sounds reasonably plausible, in a ball-park sort of way, the notation may well be off, ranging from clunky, inelegant and scored for a different instrumentation, via strangely full of holes and misallocated voices, to downright bizarre.

The vast majority of features in the application are concerned with refining the accuracy of the data extraction, and hence the coherence of the musical content and the readability of a score. It all happens at two markedly different levels of complexity.

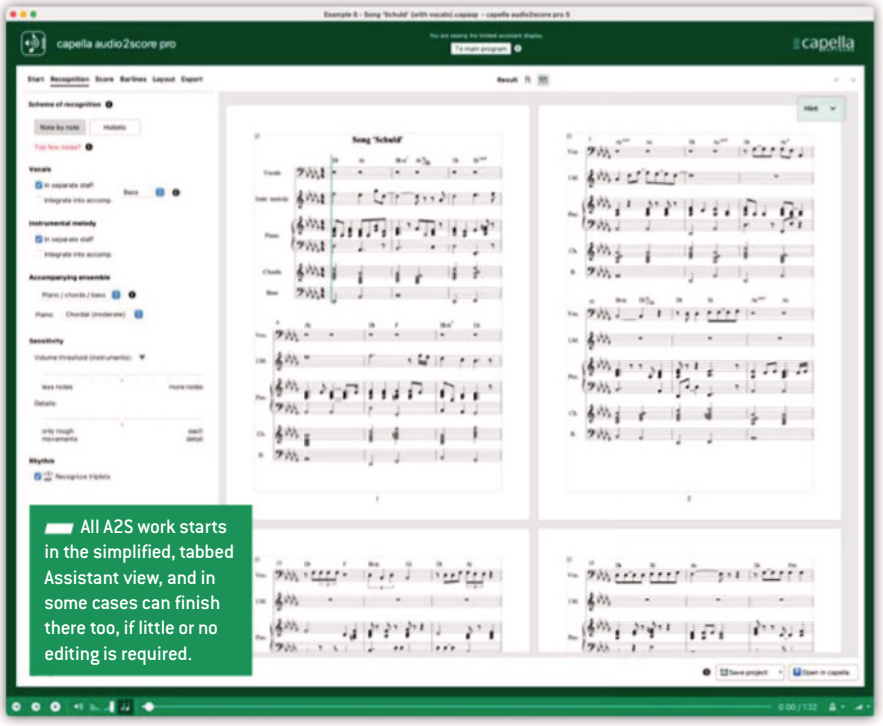
In what's known as the Assistant — where you'll begin and end your A2S journeys — you can get a lot done. Your analysis will have begun in the Start tab, and Export is the ultimate goal, but In between are several others: Recognition, Score, Barlines and Layout. There's a point to this ordering, and tweaks you make to the relatively few parameters in each tab can have a significant bearing on the outcome.

Most fundamental of all is the so-called 'Scheme of recognition'. A 'Note by note' scheme does indeed attempt to extract and represent the maximum amount of original audio file content, without adding anything that wasn't there to begin with. Interestingly, though, and perhaps surprisingly, A2S is by design not at all hung up on sticking with the instrumentation in your audio file, and indeed frequently merrily discards it.

After ingesting a piano piece, for example, choosing the 'Piano' instrumentation preset might give you a faithfully-notated result. But you could just as easily select 'Guitar' to get an instant arrangement for that instrument, notated on an appropriate octave-down single treble stave and with a playable pitch range. Conversely, you might choose to analyse a guitar performance and have A2S score it for classical organ, with a pedal part and feasible voicing and hand-stretches for a bass and treble stave above. Eleven preset instrumentations completely disregard the instrumentation in the input file, while a further seven do attempt to recognise timbre, but only within the predefined combinations of piano, strings and wind, which I think is a bit restrictive. For vocal pop music transcription, which is a new ability in version 5 of the app, some alternative options appear here, tilted towards transcription in a lead-sheet style. You get to specify the voice type, choose whether the vocal line also gets integrated into the accompaniment staves, and the type of accompanying instrument or ensemble.

A2S's alternative recognition scheme is called 'Holistic', and it's quite a different kettle of fish. This unashamedly does not look to make an accurate transcription, but instead takes an overview of your content as a whole and generates a novel arrangement based on it. It'll almost always include elements that were not in the audio file, potentially very different in style, based on seven wide-ranging preset instrumentations that start with a generic piano/chords/bass combo and end up with a nine-stave orchestra.

Many if not all Holistic scores end up largely fictional: the underlying harmonic and rhythmic content will be there, but the instrumental parts are essentially algorithmic. Piano left-hand parts frequently end up as rolling broken chords, and orchestral inner parts as alternating pairs of notes, regardless of whether there was any movement at all in the original. In the 'Piano/chords/bass' instrumentation you'll »



All A2S work starts in the simplified, tabbed Assistant view, and in some cases can finish there too, if little or no editing is required.

capella audio2score pro 5
$188

PROS
- More than just a straight transcription tool: arrangement, reduction, editing and page layout features are built in too.
- User friendly and well documented.
- At its best, it's a real time-saver.

CONS
- When the AI goes wrong, it goes very wrong.
- No recognition at all of drums and percussion, and the tech is flummoxed by distortion and some synth sounds.
- Timbre-aware scoring is subject to a limited menu of preset instrumentation only.

SUMMARY
An easy-to-use app that attempts one those notoriously difficult jobs in music: generating a musical score from a mixed, multi-instrument audio file. It works... sometimes.

» additionally get a pair of generic staves with a single-note bass line and block treble-clef chords, which can be surprisingly useful as building blocks around which you could manually build other content.

The whole premise of the Holistic scheme can seem odd at first, especially if you came along expecting the whole truth and nothing but the truth. It's immediately obvious to a notation-reading musician, though, how much generally more coherent and legible its scored output is than the Note-by-note alternative. The sacrificing of accuracy for plausibility here will definitely be the better option for some material, and for many users' needs.

To tie up this section I'll quickly mention the two sliders that adjust the volume threshold at which notes are sensed, and the level of detail in the extraction. Crank either or both up and you get more notes in your score, which may or may not help the situation.

You'll also find parameters and data display corresponding to musical key and reference pitch recognition, the number of likely harmonies in each measure/bar, and the presence and style of chord symbols in the score. Time signatures can be manually adjusted, tempo ranges quickly defined and restricted, and barlines moved by tapping along to the original audio file using the space bar. All of these are essential steps towards accuracy, especially when the AI has got the wrong end of the stick. Finally, some basic page layout elements can be controlled.

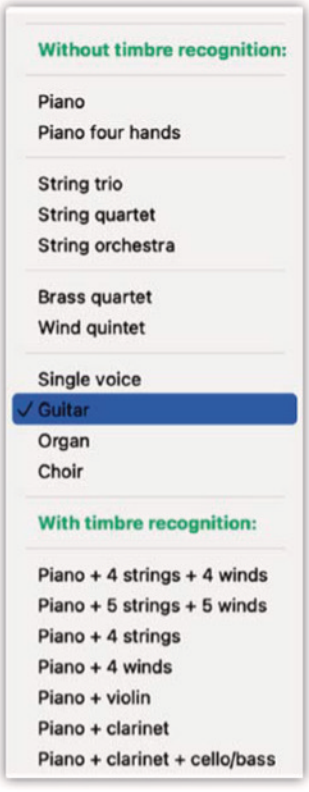
Augmented

A2S's Assistant interface is quick and convenient, but much more control lurks behind the 'To main program' button.

The main thrust here is a score view that's synchronised with a graphic representation of notes the software has found inside your audio. The Melodyne-like blobs are the 'sound tissue', with notes optionally colour-coded to their recognised timbre. The blobs are not editable, but the overlay of 'note boxes' is, creating what is for all intents and purposes a DAW-like 'piano-roll' editor. Click on a box, or a score note, and all the data associated with it appears in a panel to the right. Boxes can deleted and also dragged to change their pitch or start and end points. And for more clarity for working in detail on a complex score, whole instrumental groups or just individual instruments can be individually hidden.

To the left in this same view is a list of the instruments in the score. You can add and delete staves, and existing ones may be reordered, renamed, their clefs changed, and combinations of instruments arranged into bracketed groups in the score: all of which goes a long way towards breaking out from the preset instrumentation schemes of the Assistant. There's also support for transposing instruments here, with the underlying notation engine making the necessary changes to pitch and key signature for an instrument like a clarinet in Bb, for example.

The main program view adds some additional sophistication to the Assistant's parameters. Now you get controls for quantisation (which can be overridden on a per-instrument basis) and the tendency for individual parts to form smooth lines. In the case of chordal instruments you can specify the maximum number of notes that will exist in a chord, and the largest interval between them. Finally a 'Postprocessing' tab allows completely new notes and harmonies to be added into A2S's automated transcription, via mouse and keystrokes.



In Note by Note mode the resulting score is subject to a fairly restrictive list of instrumentations. To go further, manual editing is required.

AI IQ

The only meaningful way to assess a product like A2S is to chuck a lot of material at it and see what comes out. That's exactly what I did, and the results were... interesting.

At the risk of biasing the outcome of this review, incorrectly, I'll start by describing what happens when things don't go well. The reason being that they also reveal a lot of what A2S gets right. And in fact the situation gets rosier from then on.

Call me naïve, but the very first thing I fed into A2S was an orchestral recording: a very clear, detailed and quite acoustically dry capture of an early Classical symphony. I imagined it would be a fair fight. It had an active, full string section, plus much slower-moving four-part winds and horn, aligning quite nicely with the Note-by-note 'Piano + four strings + five winds' instrumentation preset.

The resulting score seemed close to unusable at first sight. There was a fundamental lack of fine detail (even with sensitivity sliders cranked), and all string staves were peppered with rests and the occasional individual note clearly cut adrift from elsewhere (or completely fictional). The app could not detect repeated violin quavers in a prominent main theme, and always transcribed them as long held notes. In places the metrical basis of the music, and hence the notated rhythms, had been completely scrambled. Give these parts to session musicians, in this state, and I suspect you'd have a mutiny on your hands.

But it was also interesting to see how much was correct. That included time signature, key, a lot of the meter and barline placement, most of the bass line, the »

Dos & Don'ts

A2S will tackle many different kinds of music, but its transcription skills are far from comprehensive. Most notably, there's no recognition of drums or percussion at all: it's not even attempted, and no settings will change that. So if you're still trying to get to the bottom of the Amen Break, look elsewhere.

Vocal recognition improved hugely in this latest version of the app, thanks to a new fine-tuned recognition algorithm and features purpose-built to handle them. What A2S still struggles with, though, are synthetic and distorted sounds. They're either ignored or bleed into nearby instrument staves causing all sorts of chaos. Autechre devotees probably need not apply.

Otherwise the biggest limitations include really intricate rhythm, including tuplets other than triplets for example, and an inability to track time signature or key changes in larger musical structures. That's almost never a deal-breaker though, even for the most committed prog fan, with solutions ranging from feeding audio into A2S in sections, to further editing further down the line in a more fully-featured notation application.

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Lynx

» harmony (revealed by the chord symbols), and the sections where the wind came in and out. In fact the wind writing, while not the same as the original, was often entirely plausible. Oh, and the piano stave was completely empty, as it should have been.

As a 'broad brush' depiction of the music it has to be said it was strangely successful. A lot of work would certainly still be needed to make either a usable score or MIDI data ready for playback by an orchestral sample library. However, both beat tapping and some straightforward reallocation of timbre in the main program improved things, and there are enough tools on hand to tighten things up even further if you're so inclined.

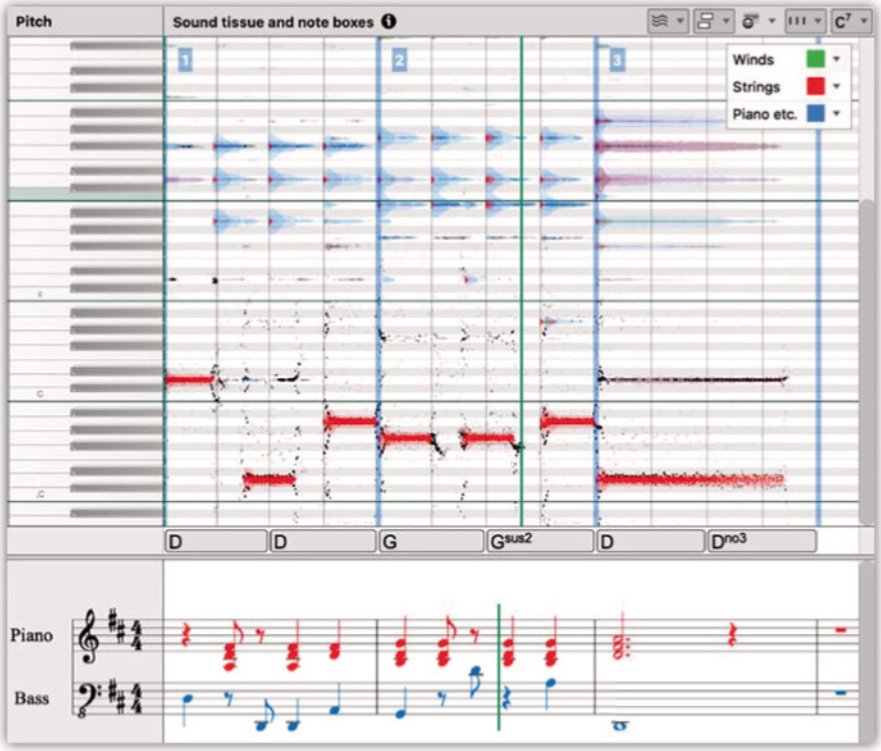
Next up, I tried some clear, non-distorted pop mixes with drums and vocals. Again, the outcomes were at first only partially successful, and a long way from what a pro-level human transcriber would come up with. And yet, again, there was still a lot of useful material that could be corrected or built on further down the line. Drums and percussion were ignored as per the app's design and didn't seem to interfere with pitched content too much. Holistic mode always did better here than Note by Note, which suffered from the cross-stave bleed I mentioned already, but the way it works takes some getting used to.

Take holistic piano parts for example, in the Piano/chords/bass instrumentation. You get to choose what goes in the right-hand part: either a simplified reduction of other instrumental parts or pretty much a copy of the Chords stave. But that quasi-arpeggiated left-hand line: wow! From a moderate 'Level of difficulty' upwards it becomes relentlessly active, and can be viciously hard to play, with leaps and figuration that constantly goes outside a typical hand span, and makes no concession for key. It's idiomatic piano writing to the extent that it sounds piano-ish, but it has the unmistakable feel of a part an AI would write, not a skilled human composer. There's little regard for elegance, such as in the frequent overlaps and crashes between right- and left-hand material.

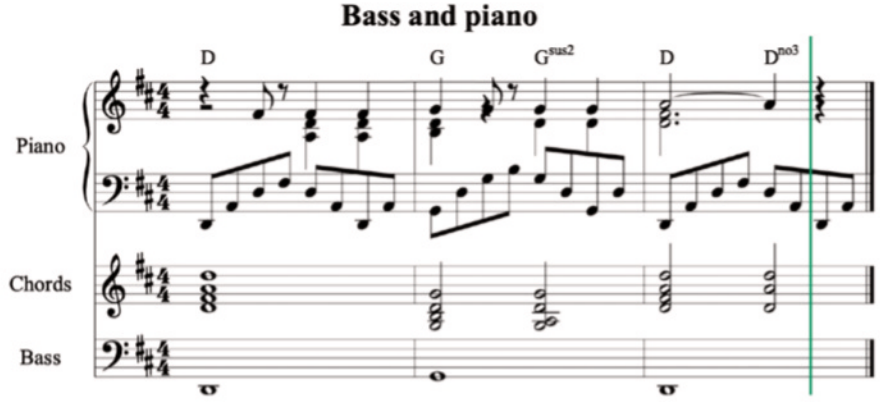
Other Holistic scoring aspects were better. In the orchestral and quartet presets you're likely to see some continuous, melodic-style parts with reasonable voice leading, and the simple but animated broken-chord accompaniment patterns are effective, in a minimalistic, neutral sort of way. Orchestral wind and brass parts are given to padding, with held chord notes evenly distributed amongst them. The application's repertoire of material is small though: those same figurations occur again and again, regardless of the material you feed in.

In the end, the best and most useful and accurate outcomes were with relatively simple material that included instrumental duos, trios and solo piano. Here, A2S would often manage pitch accuracy very well. Rhythmic accuracy was always less certain initially, but could be improved with the onboard tools. The application never recognised compound

To illustrate the sort of results you can expect from the application, I fed it an audio file of a simple piano and bass guitar ditty, notated here cleanly in Sibelius and recorded dry and hard-quantised in my DAW...



...Note by Note mode got the gist of it, but with some notable holes and miscues (despite the apparently very clear red-coloured bass timbre recognition), and had to be manually configured to display these specific instruments...



... Meanwhile the Holistic mode score performed some typical embroidery, and some simplification, for better or worse.

(shuffle/triplet) rhythmic content by itself, for me, and the resulting attempts to notate it 4/4 led to chaos. Manually entering the time signature would fix things in a blink of an eye though.

Working at this lower level of complexity, and with a few provisos in hand, there's lots to like. It accurately revealed for me a few jazz piano voicings in recordings I'd pondered over for years, and that in itself was very pleasing. I suspect it would do the same for much guitar writing too, though it's then a bit of a shame there's no tab notation option within the application itself.

Final Score

The promise of software like this, to be able to pull off transcriptions that would represent hours of painstaking work even for a professional orchestrator, is immense. It could prove to be a huge time-saver for some jobs, but has an educational potential too, revealing harmonies, note patterns, and pointing out arrangement and orchestration strategies, in audio material that otherwise has no printed or published version.

That's the dream, and there's no question the app can already transcribe some material well. Its extensive arrangement, reduction, note-generation and page layout capabilities take it to places far beyond a straight transcription tool. Various aspects of execution are excellent: it's fast, clean, easy to use, and really well documented. It's also markedly more sophisticated than tools like Melodyne or the built-in audio-to-MIDI features in some DAWs. They might appear to be similar, but most are fundamentally simpler, built to deal with separated, track-based, unmixed audio, and sometimes restricted to monophonic lines. A2S is attempting something in an entirely different order of complexity.

However, the limits of musical plausibility in A2S's output are reached surprisingly quickly, and the built-in tweaking and editing tools only go so far. Ask the app to crunch more complex mixes and it's not at all certain what you'll get in return. The Holistic mode keeps things coherent longest, and can at least reveal underlying harmonic building blocks, but does so only by discarding much of the original material and generating new. I don't have many other gripes, but one concerns the lack of warning that Main Program edits can be lost if you decide to try another recognition scheme. Another is the lack of support for 24-bit WAV files.

I instinctively applaud apps like audio2score pro, for their sheer 'have a go' bravery. The complexity of the underlying programming skill must be immense. I suspect though that truly accurate audio transcription and creative offshoots that equal what experienced human arrangers could achieve are still off by some decades and many orders of magnitude of processing power.

So don't expect perfection. However, go in with the idea that A2S will get you some of the way, and if you're willing to build on its approximations there's a lot of value here. If it can save hours of tedious donkey work — and for many jobs I think it will — then that should justify the reasonable asking price in no time at all for many users.

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ROBIN BIGWOOD

Roland RD-08

Stage Keyboard

The Roland RD-08 is a stage keyboard with hidden depths that punches well above its price tag.

Roland's RD range has been around for decades, offering various permutations of the stage keyboard concept, from simple preset-based designs to models (like the current flagship RD-2000) that border on workstation synth complexity.

The RD-08 sits at the simpler end of things, and is one of the most affordable RDs ever. It's a counterpart to the existing, slightly dearer and more sophisticated RD-88, and is clearly pitched at musicians who want easy access to high-quality, playable sounds, and not too much in-your-face complexity. As we'll see, though, there is considerable depth if you go looking for it.

Relative Dimension

Let's start with the basics. The keyboard is Roland's 88-note hammer-action PHA-4, with textured ebony/ivory feel keys and an additional resistance stage in the downstroke that mimics an acoustic piano's escapement mechanism. There's velocity sensitivity (of course) but no aftertouch.

The PHA-4 isn't Roland's top-of-the-range action, but it's a very good one indeed, with a notably lively and responsive feel. Key dip is 10mm and octave span a full-size 165mm. Mechanically it's quiet, with nothing but a damped thud on the downstroke and even less on the release. I enjoyed playing it immensely: it seems to work well regardless of musical style, though is obviously not well suited to some organ techniques and really widdly synth solos.

I'll let the accompanying photo in this review illustrate what the RD-08's control panel offers. Suffice to say, there are fewer knobs and buttons even than on the RD-88, and hugely less so than the RD-2000. The pitch and mod wheels are small and stubby too, though serviceable. Crucial stuff on the rear panel includes USB A and B sockets for thumb drives and a computer connection, audio inputs for a mic and stereo line on 6.35mm and 3.5mm sockets, and three pedal inputs. One of those is specifically for a damper pedal and works with continuous-type pedals for partial-pedalling effects; a squat switch-type DP-2 pedal included in the RD-08 box isn't sophisticated enough for that but is better than nothing. The other two pedal inputs are configurable to generate MIDI CC numbers, channel aftertouch and pitch bend, and can also step through memories/presets. All sockets are clearly labelled in white on the top panel, but from the rear only with embossed lettering on the case, which will have you reaching for your torch in dimly lit venues.

Overall build quality inspires confidence and at 13.5Kg it's easily lugged. It's worth noting the RD-08's tubby girth though, with the very shallow

Roland RD-08
$999

PROS

- PHA-4 keyboard action feels luxurious and responsive.
- Some great-sounding acoustic and electric pianos.
- Includes an extensive, broad-ranging preset-based sound set.
- Built-in speakers can be a boon in some domestic and rehearsal situations.

CONS

- Fiddly menu-driven configuration involved in anything more than basic use.
- Very limited scope for real-time control or sound programming.
- Oddly chunky.

SUMMARY

A purposeful stage keyboard with a great hammer action and some of Roland's best piano sounds, at an uncommonly accessible price. More money will buy greater sophistication, but this gets all the basics spot on.

(front to back) control surface behind the keys only accentuating the uncommon height: on my studio desk it was a hefty presence, with the surface of the white keys a full 13cm above the feet, and the total height 15.9cm. That's something you may need to factor in in some setups: for example, for a workstation with a pull-out keyboard tray, or some multi-keyboard stage stacks. The presence of built-in speakers (downward facing, with top-panel vents) helps explain the bulk, but at a rated 6W and with a rather boxy quality they're a handy convenience feature more than anything. For serious work you'll want headphones, monitors or PA as usual. Helpfully, the speakers can be disabled via a dedicated button.

Zones & Tones

I mentioned the RD-08's simplified user interface: just five sound category buttons provide access to factory preset 'scenes', and they are Piano, E-Piano, Organ and Synth, plus 'Other' for everything else. After pressing one you can explore other scenes in the same category by using the Dec/Inc buttons, but the choice is fairly limited. From the factory there are 11 piano scenes (and that includes piano/pad layers), 13 e-pianos (similar), 10 organs (mostly good-sounding Hammonds), a mere eight synths, and 20 strings, synth strings, clavinets and vibes, a few brass and synth brass sounds, and two basses. A complementary Favourites system allows quicker single-button access to sounds if you want to avoid the select-and-scroll caper. Preset switching is apparently seamless: selecting a new one won't kill the lingering sounds of an old.

It doesn't take long, surfing through presets, to discover the RD-08 is capable of playing three sounds at once, in layers (or 'zones' in Roland-speak) called Lower, Upper 2 and Upper 1. As the names suggest it's possible to have a layer (a piano and a pad, for example) and a split (a bass sound in the lower octaves) active at once. There are dedicated knobs for zone volume, as well as for an incoming mic/line signal, and buttons and procedures for enabling layers and splits are simple and intuitive.

As for further editing and tweaking, you can quite easily go into a scene and explore some alternative sounds (aka 'tones') for each layer. At first it seems only very few are available, but as we'll discover in a minute that's far from the truth. Secondary functions on the four editing knobs also then allow adjustment of allocated system-level effect levels and 'tone colour', which controls stereo width or a simple macro-style EQ, but sadly never anything like filter cutoff. Tweaked scenes are easily saved — all the factory scenes can be overwritten, and hence the category names can end up completely arbitrary — and there are a fair few empty 'INIT SCENE' slots available too.

Used like this, it's easy and quick to get down to business with the RD-08, and for the most part its sounds are of excellent quality. The main Concert Grand acoustic piano sound is super classy, modern, well-behaved, and with bags of presence and energy. The same is true of the Rhodes e-pianos, which are complex and soulful, positively inviting lengthy musical explorations. I imagine some RD-08s will spend their entire working lives producing only these sounds, and why not? They give little or nothing away to much more expensive competition, and even to cutting-edge sample libraries. »

The RD-08 Upgrade

I mentioned previously that the RD-08 has a lot in common (the entire guts of the thing, I'll hazard) with the more expensive RD-88. What it lacks are some additional panel controls, as well as some audio interface functionality and dedicated Apple MainStage integration.

Surprisingly maybe, RD-08 owners can unlock these firmware-based capabilities with a $299 software upgrade available via the Roland Cloud application that runs on macOS and Windows. *Voila!* Your 08 becomes very nearly an 88. A further unexpected bonus is that a slot opens up, into which you can load (after purchasing it, as they're not free) any Roland EXZ sound pack.

Now, this is all very well, and the optional flexibility is certainly welcome. But, call me a miser: $299 seems steep. And a rather unwelcome aspect to it all is that the Roland Cloud app can't communicate directly with the RD-08, or any other synth for that matter. Instead, it prompts browser-based downloads of purchased material, which then needs to be copied to a portable USB drive, transferred physically to the 08, and loaded that way. Someone remind me: what decade are we in again?

The RD-08's rear panel houses a USB port for computer connection and another dedicated to updates(!), a full-size MIDI out port, three expression pedal ports, quarter-inch and 3.5mm audio inputs, stereo quarter-inch audio outputs and a quarter-inch headphone port.

» Going further, Hammond organs sound like the real deal but are of fixed registration with no drawbar control. Leslie speaker rotation speed can be (and often is) mapped to the modulation wheel. Synth timbres are full and frequently fruity, and will fulfil some standard gigging duties, but they are not tweakable in real time in any practical way. Also be aware that there are no guitars, harps, non-Western instruments, percussion or drums at all in the factory scene provision.

Regional Development

As I mentioned earlier though, the RD-08 has hidden depths: there is actually a lot more here than initially meets the eye. First, it turns out that the true timbral range is barely even hinted at by those couple of dozen factory scenes. Dig only a bit further and you can find literally thousands of ready-rolled tones, in 11 banks, available to load into zones: the downloadable PDF listing them is 37 pages long! Amongst the available sounds, which are organised by category type or in arbitrarily named banks, are hundreds more good-sounding synths (some specifically referencing Jupiter, Juno, TB and SH-series Roland classics), many more organs (of tonewheel, transistor and wind-blown varieties), those missing-in-action guitars, world and ethnic instruments, and sound effects. There are scores of pianos too, though nothing else is as good as the flagship tone, and there aren't any true, high-quality 'character' options like uprights or felt variants. Tones are essentially mini-presets, and some (intriguingly) load with their own effects, independent of the normal user-facing effects. You can disable these if necessary via a menu item. Dig really deep there and you'll even find filter cutoff and resonance controls, and parameters relating to a single envelope and LFO: they can never be liberated to the front panel though.

It then transpires that the RD-08's 'upper' and 'lower' zone names don't have any particular meaning. You can use them all how you want, as all layers, as a three-way split, with overlaps, and in any order (with the lower above an upper, for example). There's no Nord-like crossfades for zone overlaps, but you can set up velocity switching, control pan position, and decide if a zone receives sustain and other pedal inputs, amongst other things. Again, it all has to be painstakingly set up in menus, but this untrammelled design adds to the flexibility enormously if you sometimes need more ambitious registrations.

In signal architecture terms the RD-08 is open-ended too. Each zone gets an effects unit of its own which can draw on close to 100 different algorithms, including respectable reverbs, delays, filters, distortions, amp sims and use-specific combos (like an amp simulator and tremolo chain for an electric piano). There are also independent three-band EQs and the 'tone colour' processors. All zones can further share a sympathetic resonance processor (meant for piano pedal resonance only, and not extending to true harmonic excitation of held notes) and a send/return type reverb and chorus, before being mixed together through a compressor/EQ stage. A fourth internal sound source is a metronome that has accessible front-panel controls. Meanwhile the external mic input gets a dedicated, menu-accessed reverb and EQ.

Taken all together the RD-08 has the potential to sound much less cut and dried than it does fresh out of the box, with scope for distorted sounds, rhythmically-sliced lo-fi textures, lengthy reverbs and more. Sadly, though, this sort of creative use is not encouraged by the user interface: the RD-08 lacks the real-time controls of a Nord Stage or of Yamaha's CP, YC or even CK series, and the menu system, though clear enough, is slow and old-fashioned. Also, while the huge provision of alternative tones is welcome, there is a needle-in-haystack quality to finding something really specific. Getting to the good stuff, through what can seem like acres of filler, will take some dedication and, in the absence of an encoder knob, many hundreds of presses of the Dec/Inc buttons.

Robust Defence

Around the time of the RD-08's release I spotted a YouTube video with a click-bait title along the lines of 'The RD-08: is this the best stage piano ever?' I can save you precious mortal moments by answering that question quickly here: no, it's definitely not. But that's not to say it's a bad one...

What we have here is a keyboard designed for affordability first and foremost. Compromises have certainly been made in some aspects of the user interface and feature set, but not, crucially, in the keyboard action and core sounds, which easily bear comparison with competitors costing even three or four times as much. The PHA-4 keybed is very fine indeed, and feels better (to my fingers) than anything Fatar-based offered by Nord, for example. And while the main RD-08 pianos sound superb, by any standards, there's still a whopping provision of other sounds, if you're prepared to explore it.

So no, the RD-08 isn't the 'best', but vital aspects of it are of really excellent quality, offering a premium experience in a budget package. That's quite rare, and I applaud Roland for concocting this combo. It might not be exotic and aspirational, but I expect the RD-08 will make a lot of musicians very happy indeed.

$ $999.99
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ALTERNATIVES

At the asking price, there are hardly any alternatives. A gaggle of home-oriented **Korgs**, **Casios** and **Yamahas** undercut the RD-08 but are limited designs that also can't match its action quality; only Roland's own PHA-4 equipped **FP-30X** can do that. Yamaha's more sophisticated and programmable **CK88** invites comparison but commands a 70 percent premium.

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PAUL WHITE

The Core 5 is the baby of Dynaudio's Core series of monitors, measuring just 190 x 353 x 265mm and weighing only 8.6kg. It has applications both for nearfield monitoring and for use in immersive multi-speaker arrays, both scenarios in which its small size is an advantage. This is a two-way active monitor with a flared, slot-shaped front port. Where more low-end power is needed, the Core 5 is fully compatible with Dynaudio's Core Sub Compact subwoofer (see box).

The power supply accepts 100-240 V AC at 50/60 Hz, and feeds twin Class-D Pascal amplifiers, with one 280W unit driving the mid/woofer, which sports the familiar Dynaudio slotted dust cap, and a further 280W available for the 28mm Cerotar Pro soft-dome tweeter. These cross over at 3.8kHz, with crossover duties handled by the internal DSP. The Core 5 delivers a bass response down to 45Hz at -6dB, and 51Hz at -3dB, which is quite an achievement for such a compact speaker. The 5-inch woofer employs Dynaudio's proprietary MSP (Magnesium Silicate Polymer) cone material, which was developed to provide a good balance of weight, stiffness and damping. Finished in a dark-grey textured paint, the cabinets appear to be made from MDF and are very solid.

The tweeter features something that Dynaudio describe as a Hexis resonance-defeating inner dome. One of the benefits of this is said to be a non-fatiguing sound. The spec on the monitor quotes the high-frequency response as being only 3dB down at 26kHz. As a whole, the monitor claims a maximum SPL of 110dB at one metre when used in a 'half space' setting.

To compensate for location relative to room boundaries, the speaker incorporates two DSP filters; there's also a setting for desk mounting, to help compensate for desk reflections. Anechoic is the setting for use when working away from desks or nearby boundaries, with other settings including Free, Wall or Corner. For use with a subwoofer there's also the option to cut the response below 80Hz using a Linkwitz-Riley high-pass filter. Further adjustments, Bright, Neutral or Dark, are available for the overall sound balance, which can be helpful either to match the sound to your personal taste, or if your room is particularly bright or dull. These settings are based on full-spectrum 'tilt' filters with 1.5dB lift and 1.5dB cut options.

All the connections and controls are on the lower half of the rear panel; a balanced XLR analogue input has switching options to set the maximum level from 0dBu to +24dBu, with a further

Dynaudio Core 5 & Core Sub Compact

PROS

- Revealing, detailed sound.
- Clarity without harshness.
- Low end is well controlled.
- Core 5 integrates well with Core Sub Compact.

CONS

- There's no straightforward way to hook up the Core 5s to the Core Sub Compact if you want to keep all your connections analogue.

SUMMARY

Both from engineering and sonic standpoints, I couldn't find anything not to like!

feature that sets a maximum output SPL while still making the best use of bit depth in the digital domain. This can be set to 116dB, 96dB or 79dB SPL (reference -20dBFS). When connecting digitally using the rear-panel AES3 XLR, the AES3 stream channel can be set with a channel switch. The internal DSP follows the sample rate of the input and can either lock to digital audio clock or be clocked from a separate BNC word-clock input. There's also an AES3 digital output on XLR and a USB port labelled Service.

Performance

Set up on my desk's bridge, which is around a metre from the wall and the windows behind, the Core 5s gave a very good account of themselves, with no need to stray from the default settings other than to switch the input sensitivity to something more appropriate to my monitor controller.

I've worked with a lot of different monitors over the years and I pride myself on being able to pick out the ones that sound accurate when tested using a wide range of material. Right away I knew that these speakers fell at the 'very good' end of the spectrum. In comparison with my 15-year-old Event Opals, which I still regard as exceedingly good monitors, the Core 5s sounded remarkably similar in overall tonal balance, other than at the extreme low end where

The Core 5 features an analogue XLR input, a digital AES input and thru out, as well as a word-clock input.

the larger Opals have a little more reach. At sensible listening levels the sound is beautifully clean and well defined, but with no sense of harshness in the highs and a real sense of spatial depth as well as stereo width. The low end sounds tight for a ported design, and when paired with the Core Sub Compact (see box), those bottom octaves fill out very nicely without the sub ever breaking a sweat. While some users are wary of using a sub in conjunction with smaller main speakers, one practical benefit is that the sub can be positioned where needed to give the flattest low-end room response without compromising the stereo imaging.

Summary

At over £2500 a pair, the Core 5s are not inexpensive monitors, and you can almost double the price by adding the sub. However, they have been designed to perform to a genuinely professional standard, and in the smaller studio, a pair of Core 5s will probably be all you need to achieve high-quality monitoring without the sub. As mixing tools, they tell it like it is, so good mixes sound 'right' and flaws are shown up clearly, just as they should be. I also found them to be very comfortable to work with for extended periods, which is particularly important in a commercial environment. The Core 5 speakers are priced singly as they are also ideally suited to multi-channel setups where more than one sub can be used if required.

$ Core 5 $1699, Core Sub Compact $2699. Prices are per speaker.
T Dynaudio North America +1 847 730 3280
E sales@dynaudiousa.com
W www.dynaudio.com

Core Sub Compact

The Core Sub Compact houses dual 'opposed' nine-inch drive units, a topography which helps cancel out cabinet vibrations and shares out the load. It is rated at 100dB SPL at 1m, with cutoff frequencies at 85dB quoted as 16Hz and 200Hz.

While measuring just 390 x 360 x 365mm, the Core Sub Compact is still a hefty beast weighing 26.1kg. Its cabinet is sealed rather than ported, which avoids the issues associated with ported cabinet resonances, and there are no speaker grilles to rattle — though this means you'll have to be careful to position the sub where the driver cones won't be exposed to accidental damage risks. As with the Core 5s, the rear panel sports a comprehensive range of DSP switch controls for position (including soffit mounting), full-range or low-pass operation, attenuation and input sensitivity. There's also left/right switching for use in a digitally linked, dual-sub system. Analogue input is on an XLR, as are the AES digital input and digital pass-through. A BNC word-clock connector is also fitted.

As with the Core 5s, power comes from a pair of 280W Pascal amplifiers, one for each drive unit. Developed to deliver powerful bass with minimal distortion, the drivers feature a hefty ferrite magnet system and a multi-layer cone utilising a rigid aluminium core that has a layer of paper damping on the rear. Dynaudio's MSP (Magnesium Silicate Polymer) material is used to form a cap on the front of the cone to tame resonances.

Having only a single analogue input means that if you want to use it that way, you'll need to handle bass management externally. The intended approach is to use the AES connection to provide a digital input to the sub, then use the sub's digital out to feed one of the Core 5s. Another XLR digital cable is then used to link the two Core 5s with their left and right switches set appropriately. The sub's bandwidth switch should be set to low-pass and the high-pass filters engaged on the Core 5s.

Unusually, the Core Sub Compact only features a single XLR input, rather than the usual stereo I/O arrangement. This is because it's intended to be used via the AES digital input, handling crossover duties internally and passing the stereo signal on through its AES digital output.

MODULAR

Vermona meloDICER & MEX3 MIDI Expander

Eurorack Modules

With the release of the MEX3 MIDI Expander, Vermona haves transformed this fascinating stochastic CV pattern generator into a MIDI-enabled chord progression instigator. We missed the chance to review the meloDICER when it was first released, and so the MEX3 gives us another opportunity to check it out.

meloDICER: the module's name pretty much nails the philosophy behind it. It generates melodies through the rolling of dice. Although more precisely, it generates patterns through stochastic randomisation.

The basic premise is that you have the 12 tones of an octave on sliders. As you push up a slider that note becomes more likely to be selected and as you push more sliders you apply influence over the possible melody and how often the notes will sound. To the right you'll find a pair of Range sliders. These let you set the octave range of the melody. That is the sum total of your melodic interference.

The Rhythm side of meloDICER is conducted via the row of four knobs along the top. The first knob sets a note length relative to the clock, from one bar to 1/32 note. Next, you have a Variation control that starts evenly at the 12 o'clock position and then introduces longer notes to the left and shorter notes to the right. The Legato control will increasingly drop gates until you remove them from the equation entirely. Finally, we have Rest, which suggests to the melody that it should hang on to the last note for a bit. With these knobs set to their lowest values, you get a straightforward gate on every time division. But once you start to exert their influence, all sorts of rhythms begin to emerge.

Without the MEX3 expansion, the meloDICER module has a single 1V/oct CV and gate output and leads you into some great melodies. To generate a suitably fabulous bass line, push up a few notes and lean into the knobs just a touch, and you're off with some pretty solid patterns. It pulls you into experimenting with range and variations, adding and removing notes or varying their importance on the sliders. The melodies and rhythms generate themselves; your interaction influences the outcome rather than dictates it. And this brings us to the Dice buttons. With the Melody and Rhythm Dice buttons lit, they will repeat the same pattern of 16 steps. While you may influence the rhythm with the knobs and change the notes, the underlying rhythm and movement of melody remain the same. If you want that to change, you hit the button, and the machine rolls the dice to create a new pattern of notes and rhythmic structure.

A roll of the dice won't give you a banger every time, but you'll quickly find something interesting and pleasing that you can mould to your liking with the controls. You can also play with the length of the loop via the spinning circle of 16 LEDs. Holding the buttons to the side, you can select the first and last note of the sequence with the encoder. Holding both, you can rotate your range through the 16-step pattern, seeking out the most interesting elements. If you'd prefer a fully random experience, then you can disengage the two Dice buttons and meloDICER will meander its way through ever-changing probabilistic melodic and rhythmic events.

The MEX3 brings this stochastic note generation into your MIDI synthesizers or DAW via TRS Type A MIDI connections. In doing so we discover that the whole module is MIDI controllable and transmits CC numbers for every knob and slider. We also find that the MEX3 is unexpectedly kicking out three-voice polyphonic patterns completely by default and without any enabling or intervention from us. This can be very confusing for anyone plugging it into a monophonic synthesizer, but once you have it pumping into a polysynth or sliding into your DAW then it becomes a differently beautiful experience to the monophonic modular CV one.

On every step you get a three-note chord derived from the same stochastic engine, but running three times to produce three notes on each step. So, it's not deciding on chords as such, it's finding chords through the random selection of three notes. You have to think about things slightly differently because all the notes you might want in a melody are not

necessarily those you want together in a chord. The results are really interesting and can range from demented pianist to synth virtuoso. However, what's not immediately clear is how we get back to a single line of melody, which is, after all, what we wanted the meloDICER for.

After a good bit of manual analysis it turns out that the three voices can be silenced through the Rest control. Turning the knob on the meloDICER affects all three voices at once and increasingly introduces rests over the whole chord pattern until nothing is playing. To leave just the first voice playing you have to use MIDI CC numbers to turn up the Rest function for voices two and three. Once you've configured your software or hardware MIDI controller with the right CCs you then have to enjoy a few hours of troubleshooting until you discover that, by default, the MIDI input is disabled, despite the flashing green activity light that would suggest otherwise. I can't quite wrap my head around why a MIDI expander that requires MIDI control as a basic function would have a factory setting that prevents it from working.

However, when it is all up, running and making sense in your MIDI and/or CV environment then meloDICER has an extraordinary ability to generate pleasing patterns with a modicum of effort. Its strength is in the ordered nature of things. From the straightforward layout to the familiarity of the outcomes, it really hits the mark. Sometimes, it feels a bit straight-laced, like a safe pair of hands that perhaps lacks a sense of adventure. But it's never been easier to write a sequence without writing a sequence.

Robin Vincent

$ meloDICER $529, MEX3 $189.

W www.vermona.com

Schlappi Engineering Nibbler

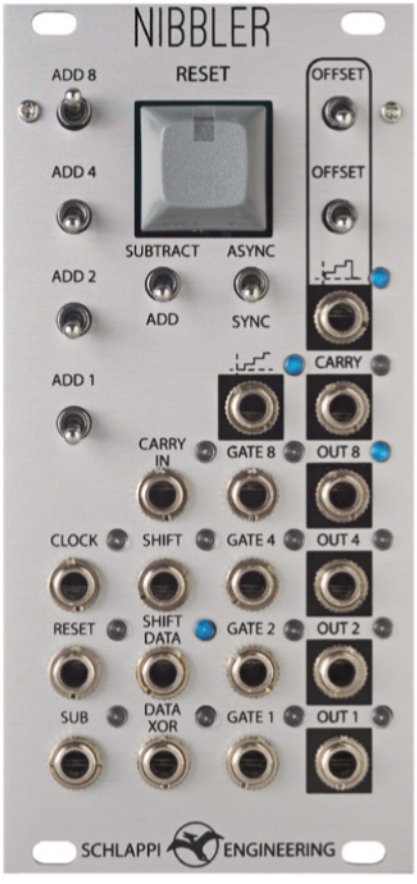
Eurorack Module

I am reminded by the Nibbler, as I was reminded by the Three Body, of how perfectly modular synthesis often rests on the boundary between art, maths and philosophy — a composite school more often associated with the great thinkers of antiquity than with electronic musicians of the 21st Century. Of course, Schlappi Engineering are far too iconoclastic to wear that kind of intellectualism on their sleeve, and literally: one of their T-shirts features the blackletter-set slogan 'We're Here To Fucking Party' and I'm yet to see evidence to the contrary.

At the core of the Nibbler is the concept of counting in binary. As the saying goes — there are 10 types of people: those who understand binary, and those who don't. If you're in the latter camp, fear not: for the sake of brevity, you need only get your head around the binary values between one and 15, which the Nibbler ostensibly counts to and from in various different ways, firing out gates and voltage values as it does so. To be honest, you don't even really need to do that: as we've come to expect from Eric Schlappi and company, the Nibbler is as smart a module as it is accessible. So it's really up to you how deep down the conceptual rabbit hole you want to go.

That concept is nonetheless an elegant and rich one: the Nibbler essentially takes the simplest fundamentals of computing and re-introduces that into the analogue domain as a musical principle. It works in 4-bit, simply meaning it has four on-or-off stages, and a maximum value of 15, or 1111, because that's where all four stages (bits) are 'on'. The four bits' outputs look a lot like a clock divider, and can almost work as such; but they behave very differently, firing in different combinations up to that maximum value of 1111, after which the accumulator will 'overflow' (firing a singular gate out of the output called Carry as it does so) and either start again from zero, or from whichever value carries over from 15 if it's counting in greater increments than one. Don't worry, it gets easier.

Patch in a clock and it'll begin counting: 0001, 0010, 0011, 0100, 0101, 0110, 0111 and so on. Each of the 'on' bits (the number 1s above) delivers a gate, and there's also a stepped voltage output which represents the sum of all the bits' values. You might have gathered by now that different combinations of gates firing off makes for an excellent, quick way to trigger multiple percussive sounds in an array of interesting combinations, while the stepped CV output behaves more like a downsampled ramp wave. There's also a second stepped voltage output that is identical to the first but with a phase offset of 0, 45, 90 or 180 degrees. Patch this with an offset to a second oscillator, for example, for a faux-delay or even a melodic canon effect.



Now, you may be thinking: 'OK, underneath all those gates is a four-bit accumulator. Who cares?' Indeed, it's quite easy on first impression to perceive the gates as simply firing off at random, if in time, and the stepped outputs little more than a clumsily quantised waveform. The answer to that question lies in the fact that as soon as you start to patch the Nibbler in, a method to the madness quickly emerges. After all — slightly dry as it might sound — mathematical divisions and multiplications are part of the very fabric of all music, both in the rhythmic and harmonic realms. With the Nibbler, then, combinations of triggers circulate and come back around at odd but also perfect moments for spontaneous and highly musical syncopation; frequencies are chewed up but retain their harmonic sense. It can be a clock divider if you want it to be, but feed it audio-rate pulses and it will spit out all manner of chiptune or dial-up modem-style sounds, its frequencies orbiting the frequency of your oscillator, not least delivering »

a variety of handy sub-oscillator options. I enjoyed paralleling it alongside an oscillator for some additional harmonic detail and movement, even using it as a quasi-bit-crushing, almost-chord-generating effects send.

And this is where the array of controls and CV inputs to influence the Nibbler's behaviour become quite exciting: four 'Add' switches allow it to count in ones, twos, fours or eights, or any sum of these. There's a Subtract switch (and jack) to have the Nibbler count backwards or forwards. A Shift input rotates the gate outputs, while a Sync/Async switch untethers incoming pulses to each bit's individual gate inputs (that's right, each bit has its own gate input) from that of the clock. There's also a giant Reset button for some performative disruption, and the rest. There's far more here than I can fit into my column space, but let me assure you that it's more than enough. At the end of the day, much like the Three Body, there's little lost in casting the abacus to one side and joyously flicking switches simply to see what happens. Slam dunk. *William Stokes*

$ $240

W www.schlappiengineering.com

Nano Modules VCV Random

Eurorack Module

It's not often that something gets pulled from the virtual realm of software emulation into the touchy-feely realm of reality, but that's exactly what Nano Modules have done with the VCV Random. Nano worked with the Eurorack emulation VCV Rack developers to bring the stock Random module to life as an actual module. In many ways it's an unremarkable and unassuming module that's just one of many random voltage generators. But once it's hardware, it immediately takes on a more focused and deliberate vibe, precisely because it's now one of only a handful of sources of randomness in my rack.

The layout is beautifully ordered, with four glowing sliders, a row of knobs, 11 patch sockets and one switch. The controls look pretty obvious, and it is refreshingly simple to operate. The basic idea is that it generates random voltages over which you have a bit of control, and that's about it.

However, the control you do have is so elegant and inviting that you may end up spending far more time here than you thought. It will be of no surprise to learn that the Rate slider controls the speed at which random values are generated. The Probability slider decides how likely a new value should occur on each clock pulse. The Random slider suggests how far the new one should leap from the previous value. And Shape smooths out the journey from one value to another. Each slider is CV controllable externally or by feeding back the values from the Linear output, so it's completely possible to set this module into a cycle of evolving randomness within ever-shifting parameters.

At the bottom of the module, you will find four very similar outputs. Each one contains the same random values, but they present them slightly differently. Step is where each new value is leapt to instantly, resulting in a familiar stepped, sampled and held output. Increasing the Shape control adds additional steps between the steps creating something approaching a glissando between values. The Lin jack is a linear output that is the same as stepped when Shape is down but increasingly slews between values as Shape is increased, until it takes a whole clock cycle to reach the next value. Exp shifts between each value with an exponential curve. As Shape is increased, the curve moves towards being linear. And finally, Smth gives a smooth transition between values with the Shape dictating how quickly the transition occurs.

If you view the outputs as traces on an oscilloscope, they ride pretty much on top of each other and I wondered whether all the real estate given over to these individual outputs was really worth the difference. But as I've spent time with it, I've come to see them as different flavours of change. In the density of a large patch, you can lose the saltiness, but with a little bit of focus the way each output plays with a filter cutoff is completely enchanting.

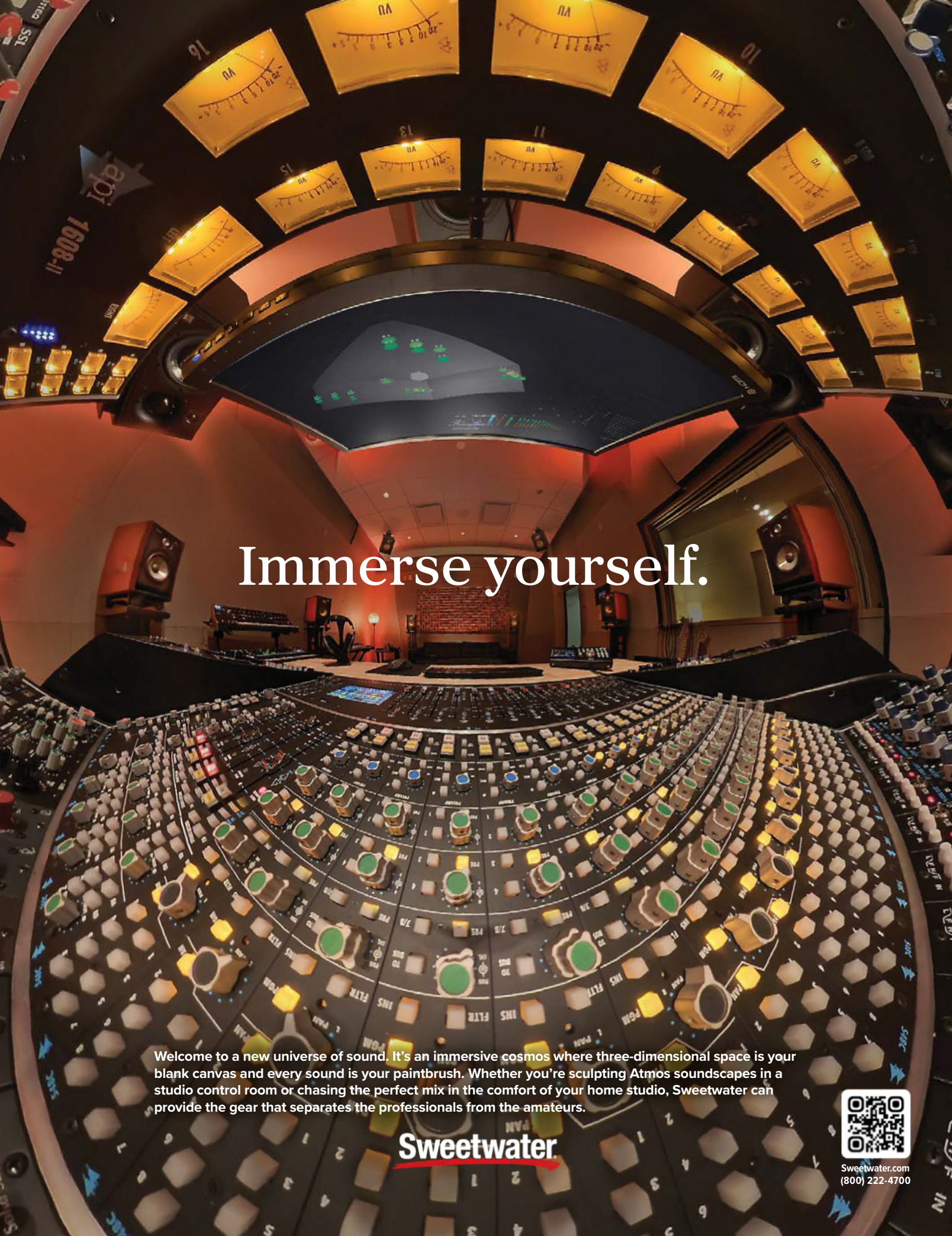
It's tempting to try to use all four outputs all at once simply because they are there, but I think I've worked out that that's not really the point. It's better to find the right flavour for the job in hand. However, if you use one output through a quantiser to run a sequence and another to work the filter cutoff, you end up with something approaching filter key tracking.

There's more; you can plug an external signal into the sample and hold circuit to give it another source of values. You can push the Rate into noise that sounds coloured and filtered by the other controls. Incoming gates can be converted into triggers or you can use Probability to put some laziness into an incoming clock pulse or to vary the triggering of envelopes and percussion hits.

I'm usually in the habit of letting randomness do its own thing in a patch. But with VCV Random, the taming and shaping of the values is immensely satisfying, whether it's melodic or modulation. While it is simple, there's enough flavour here to keep any patch coming back for seconds. In a neat circle of happenstance, the Nano version of the Random module is available for free on VCV Rack. *Robin Vincent*

$ $189.99

W www.nano-modules.com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MOTU UltraLite Mk5

USB Audio Interface

MOTU's newest UltraLite interface is jack of all trades... and master of most of them!

CHRIS KORFF

Since the Mk1's release in 2006, the UltraLite has always been something of a misnomer. While they've always 'only' been half-rack interfaces, UltraLites have also been pretty rugged affairs, and so not at all light in terms of weight. They've also managed to cram a remarkable amount of I/O into their compact chassis, so they're not light on channels either. In fact, they've arguably encroached on the 'full-fat' interface territory occupied by MOTU's own 828s, and the latest Mk5 is no exception.

MOTU UltraLite Mk5
$649

PROS
- Impressive I/O count.
- Exemplary audio performance.
- Powerful DSP mixing via easy-to-use CueMix software.
- Remote control over WiFi.
- Excellent value for money.

CONS
- Not too ergonomic for desktop use.
- A dedicated phone app for monitor control would be handy.

SUMMARY
A rugged, high-quality, great-sounding and remarkably flexible interface that crams a huge amount of features into its half-rack format.

The UltraLite Mk5 has been around for some time now, but while reviewing it last year, I contacted MOTU to mention a few limitations to the software control side of things, and they replied to tell me that they had a major update planned that would address my wish-list and add more features besides — some of which come directly from the recently launched next-gen 828. I'll cover those features later on, but first, the basics.

Let There Be Lite

The UltraLite Mk5 is a USB 2.0 interface. The USB 2 standard has more than enough bandwidth for the Mk5's I/O count, and its use here has allowed MOTU to make the Mk5 class-compliant — so you can connect it to a macOS, iOS, iPadOS or Android device without needing any drivers. (On Windows, as ever, you need to install ASIO drivers to use it with most DAWs.)

The I/O on offer are many and varied. You get two mic/line/instrument inputs on combi sockets on the front, plus a stereo headphone socket. Round the back, you get six further analogue line-level inputs, a whopping 10 line outs, a pair of optical I/O (for ADAT or S/PDIF), a pair of S/PDIF coaxial I/O, and full-size MIDI I/O. Completing the socketry are the USB-C port and a 15V DC power inlet. Speaking of power, unlike the first three generations of UltraLite, which had FireWire connectivity and could be bus-powered, the USB Mk5 must be connected to a PSU. Not much of a problem if you're using it with a computer or laptop, where you'll probably be near a mains socket anyway, but it might be a frustration for people who use tablets and the like for mobile recording.

Twist It, Push It

The front panel offers hands-on control over the two combi inputs, the headphone output and the monitor level, as well as power on/off (via a long press of the main volume encoder). Phantom power and an input pad are engaged individually for each preamp using buttons along the bottom of the front panel, which feature blue and red backlights for the pad and phantom power respectively. Encoders above those buttons adjust mic gain — up to a massive +74dB. The third encoder adjusts monitor level or, after a short click, the headphone output. The two gain knobs also provide access to a housekeeping menu: a click on the channel 2 encoder brings up options on the screen for changing the sample rate (all the usual rates are available, from 44.1 to 192 kHz), choosing a clock source (internal, optical or S/PDIF), the optical I/O format (ADAT or S/PDIF, independently for input and output), engaging 'thru' behaviour on the MIDI output port, setting the display time-out time (30 seconds, half an hour or off), displaying the serial number

and firmware version, and performing a factory reset. The channel 2 encoder scrolls through the options and selects/changes them with a click, while channel 1's encoder lets you click to go up a level.

When the display is timed out, a solitary pixel slowly bounces around the screen in *Pong* fashion, to let you know the unit is still on. In its default state, though, the screen functions as a meter bridge, showing the levels going into and out of all the analogue I/O, alongside the current sample rate, clocking status, and digital I/O activity (including MIDI). It's clear, bright and easy to read, and the menu is easily navigable.

CueMix

Like other MOTU interfaces, the UltraLite Mk5 is graced with built-in DSP mixing abilities, which are configured and controlled using the CueMix software. The Home tab duplicates the front-panel functionality, with knobs for monitor/headphone output level and preamp gain, and buttons for pad and phantom power. There are also buttons here to enable direct monitoring, whereby each analogue input can be sent directly to either or both of the monitor and headphone outputs, avoiding the computer's audio buffers and providing near-zero-latency monitoring.

The Device page provides all the same clocking, sample rate and digital I/O options as the front-panel menu, as well as drop-down menus for setting the input meters to pre/post fader, engaging a low-latency mode (Windows machines only), making the stereo output mixes from the computer available to the input mixer (so any mix you set up for a pair of outputs on the UltraLite will get its own fader within each output mix tab), and for changing the buffer size. This can be set as low as 16 samples, which at 44.1kHz yielded a commendably low latency of 1.2 and 2.0 ms (input and output, respectively). At a less CPU-taxing 64 samples, I/O latency was still a perfectly serviceable 2.2/3.7 ms.

There's a software loopback input, for recording the stereo output from other software into your DAW, and you can choose whether this appears to your computer on channels 9+10 (the default) or 1+2. The latter option is intended for fussier bits of software that only 'see' the first stereo input on your machine.

You also get the option to make any of the analogue, digital or loopback I/O visible to Windows. For most, I imagine the default option of having only inputs and outputs 1+2 available will do fine, but if you want your Zoom calls coming out of ADAT channels 7+8 and Cortana to respond to the S/PDIF inputs, or whatever, you can make it so. Finally on the Device page, you can save and load presets (which store the entire state of the UltraLite), view the interface's serial number and current firmware version, and update the firmware and CueMix app.

> **"The excellent converter quality and monitor switching make it a more than valid choice for in-the-box mixing and mastering..."**

Next is the Input tab, which again offers control over phantom, gain and pad settings on inputs 1+2, and which also lets you choose whether the signal sent to your DAW comes pre or post any DSP processing (more on which below). You can also add up to 20dB of digital gain to any of the line inputs here too, as well as flip the polarity on any/all of the analogue inputs, and set the line-in DAW feeds to pre/post DSP. At the bottom of this window you get level metering for the S/PDIF phono and optical (S/PDIF or ADAT) inputs.

Moving on, the Output tab provides output trim controls for all the analogue outputs (in pairs), plus control over the headphone and monitor level. The monitor section at the top additionally lets you create a Monitor Group. By default, outputs 1+2 are controlled by the monitor level knobs (both in the CueMix software and on the front panel), but you can select any other output to also be governed in the same way. So, you could connect a subwoofer to output 3 and have your monitors and sub all be controlled by the front panel, or even have level control over a surround setup or a 10-speaker Atmos array. That's pretty significant, and I think makes the UltraLite Mk5 the smallest and cheapest 'Atmos-ready' interface you can buy. And, because the headphone output is independently addressable (via a menu just below the Monitor Group section), you could even route a binaural downmix to that while controlling all 10 of your Atmos speakers from the front panel. It's not as 'Atmos-native' as the Audient ORIA reviewed elsewhere in this issue, but if you needed to set up a portable (if such a thing exists) or temporary Atmos rig, the Mk5 would have you covered.

At the bottom of the Output window, you can choose which of your CueMix mixes comes out of the phono and optical S/PDIF outputs, and also what comes out of the TOSlink port when you're using it for ADAT. You can either present the eight outputs to your DAW for use with an external D-A converter, or feed it from the first eight line inputs — a setting it remembers when you disconnect the Mk5 from the computer, meaning you can use it as a standalone ADAT converter. (As ever, the ADAT I/O can carry eight channels at base sample rates, or four at 88.2 or 96 kHz.) Finally for the Output options, you can select the loopback source in much the same way as you can the headphone out: it can mirror any of the hardware outputs, or be fed from any of the CueMix busses.

Making Mixes

The remaining tabs are essentially duplicates, and let you set up individual stereo mixes for each of the analogue outputs. All of the inputs (including digital) are fair game for mixing, and all of the analogue (but not the digital) inputs have access to a four-band parametric EQ and »

The rear panel hosts no fewer than six line inputs (in addition to the front-panel combi sockets), and a whopping 10 line outs, plus digital I/O and full-size MIDI ports.

» a compressor, with the two front-panel combi inputs also getting a gate. There's a three-band parametric EQ available to each line output pair. On both the input and output EQs, the uppermost and lowermost bands can be set to shelving or peaking types, with the lowest gaining an additional LPF mode. The slopes or Q of the shelving/high-pass filters can't be adjusted.

Additionally, there's a built-in reverb that can be accessed by all of the inputs, including the ADAT and S/PDIF ins. Rather than getting a dedicated aux on each channel strip, this has its own tab at the bottom, for adding reverb in the 'sends on faders' mode. There's only one global reverb mix, but as on the other mix tabs, you can send varying amounts of that mix to other outputs. The reverb parameters themselves can be adjusted by clicking the little 'decay time' window on the right of the Reverb tab. This opens up a plug-in-like GUI, with all the usual parameters (pre-delay, decay, damping, stereo width), plus a choice of small, medium and large spaces.

In short, then, you can do some pretty sophisticated mixing using only CueMix. Need to send different foldback mixes to different performers while recording? Easy. Want to send a click track from your DAW to a drummer's in-ears, while sending backing tracks and soft synths to front of house? No sweat. The CueMix interface is neat and uncluttered, and gives you the ability to rename channels, and to hop to the next/previous channel's EQ and compression window using your keyboard's arrow keys, which makes setting up a mix a quick affair.

Specs

In terms of the Mk5's technical performance, the specs are impressive. The mic inputs claim an EIN of -129dBu (A-weighted), THD+N (unweighted) of -115dB, and a dynamic range of 115dB (A-weighted). The instrument inputs are only 4dB noisier, and offer the same +74dB maximum gain as the mic preamps. The front-panel jacks present a 1MΩ impedance to unbalanced sources.

The line inputs boast an even more impressive dynamic range of 120dBA, and a similar THD+N figure of -113dB, while the line outputs claim a quite remarkable dynamic range of 124dBA. That puts the UltraLite Mk5's D-A performance in the company of products like the Focusrite Clarett+, and even Prism's flagship multi-channel converter, the ADA-128. In other words, it's well into 'mastering quality' territory.

Like those of other MOTU interfaces, the line-level inputs and outputs are DC-coupled, meaning they're capable of sending and receiving 0Hz signals (ie. direct current). This means you can use them to send and receive control voltages to and from modular synthesizers, using any software that allows it. Some DAWs can do this using their own bundled plug-ins (namely Bitwig and Ableton Live), and there's a host of third-party options too, from Expert Sleepers and NI among others.

What's New?

Earlier this year, MOTU released a major firmware update for the Mk5, along with a new version of CueMix. Updating both was simple; the firmware update took seconds and could be performed from within CueMix, while the CueMix update necessitated a reinstall and a reboot. So what do you get for your trouble?

If you're anything like me, you'll have noticed an omission among the monitor control facilities in CueMix. Being able to create a Monitor Group is all well and good for subwoofers and surround systems, but what if you've got more than one stereo pair of monitors? The answer, thanks to the new update, is AB mode. Click the new AB ON button in CueMix (it's visible in both the Home and Output pages), and it designates outputs 1+2 and 3+4 as monitor output pairs, which you can switch between using the A and B buttons. You can also engage them both by hitting the Both button.

Also new is a talkback feature, whereby you can select any of the analogue inputs as the talkback source, and route it to any/all of the analogue outputs. You can also set the talkback level, a dim amount, and choose whether the Talk switch is momentary or latching. And, of course, whichever input you choose as the talkback source still gets to use its EQ and dynamics processing.

Finally, and perhaps most significantly, the new update allows the UltraLite Mk5 to be

The Device page in CueMix. The monitoring section at the top has been recently added, and lets you set up A/B monitor switching.

controlled remotely, from any computer or mobile device that's connected to the same network as the UltraLite host. In my case, that meant I could control the UltraLite that was plugged into my Windows machine using the CueMix app on my iPhone (the same app I'd also use if I wanted to connect the Mk5 to my iPhone directly). All you need to do is enable Network Control in the Device tab in CueMix on the host system. You can set an access password here (to stop rogue states ruining your foldback mixes), and it's also possible to control multiple UltraLites from the same device. You can give your UltraLite a unique name in the Device tab, to make it easier to keep track of which interface you're controlling.

The UltraLite Mk5 hosts a fully featured DSP mixer, with EQ and dynamics processing available to every analogue input.

If you've been interface shopping lately, you may have noticed that the new AB mode, talkback and remote control facility are remarkably similar to those offered by the newly launched 828, and in fact MOTU say that they were originally developed for the 828 before being ported to the UltraLite. That such useful features have been added to a product years after its initial launch bodes well for the UltraLite's longevity and continued support, I think.

In Use

In practice, using the UltraLite Mk5 is a pleasure. The encoders feel sturdy and affirmative in action, the screen is bright and clear, and the menu options are self-explanatory. The physical format lends itself well to being housed in a rack, but perhaps less so for desktop use. Given the DSP and mixing on offer, and now the monitor control features, a 'desktop wedge' version with buttons and a larger volume encoder on the top would be great — but would also render it unrackable, so I can see why they've opted to put all the controls on the front.

Test Spec

Windows 10 Professional. Core i7 4790K CPU, 32GB RAM.

Speaking of monitor control, while the AB mode is most welcome, I'd also appreciate mono and mute switching. MOTU say that these should be simple to implement, though, so another firmware/CueMix update might soon oblige.

The CueMix mobile app worked perfectly on my iPhone 14 Plus for controlling the UltraLite remotely, with changes on the hardware reflected instantly in the app and *vice versa*, although the app was obviously designed with iPads in mind, so a fair bit of scrolling is necessary to find the channel or parameter you're looking for, and landscape orientation is pretty much a must. That said, adjusting EQ and compression settings worked well using all the usual tap/drag/pinch gestures. Finally on the iOS app, CueMix does let you use your phone for level control and monitor switching, but on the iPhone at least, it isn't especially touchscreen-friendly. A redesigned monitor control section, or perhaps even a dedicated monitor control app (with mono and mute switching!) would completely eliminate my need for hardware monitor control. Again, when I put this to MOTU they said they liked the idea, so perhaps we'll see it implemented in the future.

As for who the UltraLite Mk5 might be for... Well, the excellent converter quality and monitor switching make it a more than valid choice for in-the-box mixing and mastering, while the ADAT I/O means it can be expanded to cover hybrid hardware mixing setups and multitrack recording scenarios. Its 10 line outputs and Monitor Group functionality make it capable of playing 7.1.2 or 5.1.4 Atmos (or any of the common 2D surround formats), and the DC-coupled outputs could see it equally happy nestled among a tangle of modular patch cables. The ability to connect to mobile devices as well as computers only adds to that versatility, as does its compact form factor and WiFi control option.

One indication of who MOTU think the UltraLite Mk5 might be for is the fact that they sell a rackmounting kit for housing two UltraLites in 1U. But why would you want two identical audio interfaces? Well, if you're the musical director of a live show that Can't Go Wrong, a redundant setup is pretty much a must, so having a second UltraLite to fall back on should the first fail would be a significant comfort. And thanks to CueMix's ability to control multiple interfaces from a single device, switching to your backup UltraLite would be the work of seconds.

In short, then, unless you simply need more I/O than the UltraLite can provide, it should fit almost any workflow imaginable.

$ $649.95
T MOTU +1 617 576 2760
W www.motu.com

Electro-Harmonix POG Pico

Polyphonic Octave Generator Pedal

The Electro-Harmonix POG, which stands for 'polyphonic octave generator', has been around for a number of years and has spawned both Nano and Micro versions, but now its essential POGishness has been condensed into the even smaller Pico-format enclosure. This diminutive red pedal follows the four knobs and one button format of other EHX Pico pedals. The bypass is electronically buffered, and if using a pedalboard PSU you need to know that the current requirement is 100mA — though a separate PSU is included in the price.

Separate volume knobs set the levels of the octave-up, octave-down and dry components of the sound, while the fourth knob, Tone, has three different modes that you select using the Filter button. The colour of the LED indicates which mode is active, showing green for Tone mode, red for Low-pass mode and orange for High-pass mode. In Tone mode, you get a tilt-like EQ with a flat frequency response in the centre, then as you turn the knob clockwise you get more treble while the low end gets pulled back. Go the other way and the bass comes up while the treble comes down. Tilt EQ is applied to all three voices in this mode. The low filter works below 300Hz while the high filter works above 800Hz. In Low-pass filter mode, the filter has a resonant character; the Tone control attenuates the highs in the

sub and octave-up voices, leaving the original sound unprocessed. And no surprise for guessing that High-pass filter mode cuts lows in the sub and octave-up voices, again leaving the original dry sound unchanged.

The key thing about the POG technology is that the pitch-shifting works reasonably smoothly on chords, not just monophonic lines, though it is smoother still on single notes. It may not sound quite as pristine as some of the 'big gun' studio rack units, as polyphonic sounds can still exhibit some of that tell-tale shimmer, but it can still sound really impressive when mixed with the dry sound. Play monophonic parts and the pitch-shifting is rock solid, with none of that shimmer.

Mixing in just a little of the upper octave creates a viable faux 12-string guitar sound, while mixing in just the sub octave is great for big, doomy riffs. Mix all three voices and you can achieve a fat organ-like sound, which sounds even more convincing if you pass the result through a rotary simulator pedal. Speaking of which, I wonder if the EHX Lester rotary effect might eventually make it into the Pico range?

If octave sounds are a big part of your sound, then one of the more comprehensive POG pedals might be a better choice, but if you just want to throw in the occasional bit of octave magic without sacrificing much pedalboard space, then the POG Pico fits the bill perfectly: while it's scaled down in physical size, the sound is still as big as ever and those three filter modes offer a useful amount of tonal shaping.

Paul White

$ $240.
W www.ehx.com

Instant Audio BassRift

Bass Instrument Processor

Marketed under the United Plugins umbrella, Instant Audio's BassRift is a split-band processor designed to modify bass (whether synth or guitar) sounds in creative ways. It supports all the usual Mac/Windows formats including AAX, and it's authorised using a personal licence file that allows it to be run on all the owner's machines.

Essentially, the user can set a frequency split point below which the signal goes to a section called Safe Bass, and this runs in parallel with a processing chain that treats only the frequencies above the split point. As always with a crossover, there's a little overlap between the two bands, and the user can set the slope of the crossover (6, 12, 18 or 24 dB/octave) as

well as adjusting the levels going into each band. Solo buttons allow either band to be auditioned separately.

The Safe Bass path, shown in blue, does offer some processing: there are controls for Compression, Enhance and Mono, the latter reducing the stereo width for the low frequencies. Meanwhile, frequencies above the split point, shown in red, pass through three stages termed Flow, Space and Heat. By dragging their blocks in the diagram below, their position in the signal path can be changed. Each processor has its own bypass button and a dual-concentric knob, whose inner section adjusts the amount while its outer section governs Rate, Time or Dirt for the relevant section. Effect options are chosen by clicking directly on the name boxes around the knobs.

Flow offers variable amounts of Phase, Chorus, Flange or Wobble with a tempo-sync option, while Space offers Comb, Delay, Verb or Reverse, again with a tempo-sync option. Heat serves up distortion in the form of Tube, Crush, Trash or Fold. When the two paths are recombined in the Merge section, there's a valve-style three-band EQ, a separate De-mud control that targets boxy-sounding frequencies, and sliders for high- and low-pass filters. The Maximise knob applies

multiband limiting, and additional controls allow for the random generation of presets, hard clipping that can be applied to the output, and AGC, which endeavours to match the output's loudness to the input. The amount of oversampling can be set by the user and there's also input and output level metering.

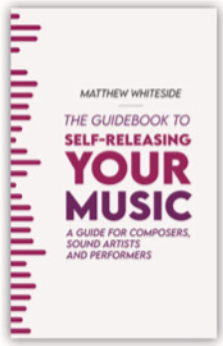
There's a good range of presets but cooking up your own treatments is definitely more fun. I found that the comb-filter option works best with the Space section placed after the Heat section, so that it has a harmonically rich signal to work on. There's a frequency read-out for the comb filter, though by double clicking on it you can choose pitches by musical note if you prefer. The types of added dirt run from warm and valve-like to wavefolding, which can add some spectacularly transformative harmonics, often adding reedy overtones. The modulation treatments in the Flow section can also have a dramatic impact, with Phase making one of my bass guitar patches sound like a Tuvan throat singer!

Bass guitars can be made to sound like synthesizers, while basic synth sounds can take on an added layer of complexity. The two-band processing ensures that low frequencies can be kept clean and punchy, while the higher harmonics are manipulated to add character. The plug-in can also work well on some midrange sound sources, but bass is where its focus lies — and very effective it is too! *Paul White*

$ $85 (discounted to $16 when going to press).
W https://unitedplugins.com

Guidebook To Self-releasing Your Music By Matthew Whiteside Book Review

In a surprisingly on-target example of online marketing, ads for Matthew Whiteside's *Guidebook To Self-releasing Your Music* began to appear in my Facebook feed a few weeks ago. And as somebody who both finds the commercial side of the music business interesting, and who has a slowly growing collection of almost finished music that someday it would be fun to set free, there was no question that I wanted to read it. So I requested a review copy from Matthew (appropriately, the book is self-published, with an editor paid via crowdfunding) and it arrived the following day. And, cutting to the chase, it's really great. Both an interesting read and incredibly useful if self-releasing is your thing.



Matthew is a professional musician and composer working primarily in the contemporary classical genre, with numerous standalone works, as well pieces for film and TV, to his name. He's carved out a successful niche for himself. He self-funds and self-releases most of his music and he has clearly over time become extremely knowledgeable and proficient at the process, so the information contained in the book is very much written from a "been there and done it" perspective. The word 'guide' is absolutely spot on: it's a book that could very effectively hand-hold you through a self-release process, although at the same time, it's not overly prescriptive. It offers suggestions, principles and concepts, but at the same time accepts that you might need or want to do things slightly differently.

It's not a huge book and I found it a pretty comfortable read over a few days. Matthew's writing style is engaging and approachable and he explains some pretty complex concepts, particularly in the sections on Rights and Registrations, very effectively. Along with those sections, the book covers the whole creative process from choosing people to work with, be they other musicians or producers, through budgeting, fundraising, recording, post-production, registering rights, distribution, PR, advertising and using social media. Some sections are UK-centric in that they describe, for example, the music rights and registration process in the UK, but even so, the basic principles Matthew describes hold true or have equivalents in other territories. The book ends with a fascinating explanation and analysis of the music streaming business, and if you didn't really appreciate just how broken is the streaming business model, prepare to have your eyes opened and mind boggled.

The sections in the book on recording, mixing and post production land in territory where we *Sound On Sound* readers like to think we know every hill, dale and shortcut. However, firstly, it does no harm to read an alternative perspective on things that we believe we already know and, secondly, Matthew's work predominantly resides in a genre where musicians play from scores using acoustic instruments, and the disciplined style of recording that demands is many miles away from, say, endlessly scrolling through reverb plug-in presets or deciding just how dirty that strummed D minor should be. It's a discipline that I suspect many of us rather less professionally trained musicians (I may well be lecturing myself here) might find useful to borrow from.

Matthew doesn't pull any punches in describing just how much hard work, organisation and commitment is involved in making a professional and successful job of self-releasing, but in explaining the processes so clearly he makes it all seem more feasible than daunting. If self-releasing is on your list of things to do, Matthew's book is very highly recommended *Phil Ward*

$ eBook £10 (about $13), softback £15 ($19).
W www.matthewwhiteside.co.uk/product/self-release-guide-book

Iconic Instruments SP140

Reverb Plug-in

The SP140 reverb is the first effect plug-in from US company Iconic Instruments. As you can probably guess from its name, it's based on the EMT 140 plate reverbs that were in common use in studios before digital reverb eventually replaced most of them. Plate reverbs have a neutral character, by which I mean that they don't suggest a particular type of acoustic space, and their reverb tail also builds up in density much faster than a typical room or hall so there are no obvious early reflections. This tends to makes them a good fit for many types of music production that require a 'flattering' ambience. The decay time of physical plates was adjusted using dampers, and in this SP140 reverb plug-in, that's achieved using a vertical meter pointer. VST and AU formats are supported as are Mac and Windows hosts, but there's currently no AAX version. Authorisation is via a serial number that's issued on purchase.

Just as with a real plate (whose mechanical dampers tended to take out the high frequencies more than the lows), very short decay times sound a bit 'dongy' »

» so that's one tick in the box for authenticity! Unlike a real plate, though, the SP140 has a few handy tricks up its digital sleeve. One such nicety is that the reverb density can be changed (there are five settings) and another is that you can call up five different plates without having to call a removals company! There's also a helpful concession to the stereo age: the input Send Width can be adjusted so that you can mono the input for a classic plate sound or widen it to bring more reverb separation to stereo sources. The reverb output width is also adjustable, along with its pan position. Large knobs control the wet/dry mix and the amount of drive, which is useful if you want to add a bit of warmth or dirt to the reverb. This can be switched pre or post the reverb engine and has adjustable high and low filters to tailor the sound.

With a traditional plate, there was no pre-delay facility — if you wanted pre-delay, you had to tie up a tape machine — but here it is built in with a 200ms maximum delay time. To further shape the reverb there's a three-band EQ. This has adjustable frequencies and slopes for the high-and low-shelving filters along with a parametric mid that offers full control over frequency, Q and cut/boost.

There's a decent range of presets here to get you started, but setting things up from scratch is really easy anyway. The five different plates each have their own distinct coloration so, in combination with the variable density, adjustable decay time, comprehensive EQ and the ability to add a bit of dirt, there's actually quite a lot of scope for shaping your reverb sounds. At maximum density you get the classic plate smoothness whereas at low density settings, the sound becomes a little more spring-like (without the unwelcome twanging!). If you share an affection for plate sounds but would like a little more freedom in shaping the end result, this plug-in a is a good choice and it's not unreasonably priced. *Paul White*

$ $59 (discounted to $39 when going to press).
W www.iconic.nyc

Soundevice Digital Randomachine

Creative Randomisation Plug-in

Available through United Plugins, Soundevice Digital's Randomachine introduces adjustable random processing elements that can be useful in adding variation to an otherwise monotonous part, such as a drum loop or pad sound. There are six separate 'sound changers' and each time Randomachine is triggered, multiple parameters change for each processing element. Triggering can be in response to audio input transients, DAW tempo, or MIDI. In Transient mode there's a sensitivity adjustment, while sync'ing to tempo allows different note-value increments to be set, and MIDI mode is triggered by incoming MIDI notes.

As with other plug-ins available from United Plugins, authorisation is through a personal key code that allows the software to be run on multiple machines, updates are always free and all common plug-in formats are supported including AAX.. An intelligent CPU-saving sleep mode is activated when no audio is present. There's external side-chain support, and Logic Pro users are catered for with a MIDI Controlled Effect version that allows audio to be fed in from a side-chain. Bypass is also click-free with no latency issues. Processing is 64-bit and runs at any sample rate to in excess of 192kHz. Various oversampling settings are available and, as usual, the higher the oversampling, the greater the CPU load.

For each sound modifier there's an amount control that sets the degree of randomisation, and if you set this to zero you'll prevent any variation of that parameter. Pitch introduces random pitch changes and Formant varies a formant filter. Pan targets left/right panning, while Ambience adjusts the time and space parameters of the in-built reverb. Time creates timing changes that can either be drastic or subtle, while Distortion varies the degree of saturation. Double-clicking the control knobs provides a finer degree of setting via a pop-up numerical keypad.

The controls for each processing element are arranged as knobs around a central zone, and this shows the MIDI notes when MIDI control is selected. Each knob shows its numerical value and the dots around the edge move when triggering occurs. The central region also pulses to show triggering. In Transient mode, the sensitivity value is shown and this can be changed by click-dragging in the central area. In Sync mode the note value is shown and can be changed from a pop-up menu. The triggering mode is set at the bottom of the GUI and presets can be accessed at the top of the GUI.

Having control over the intensity of the individual random stages certainly helps breathe life into otherwise static drum loops. It can also be used to ring the changes for arpeggiated sounds or other musical element that needs a touch of organic variation, though I found it most useful on drum loops and percussion parts. Adding some formant-shift, maybe a tiny amount of timing variation and a hint of variable distortion really does break up the monotony of repetitive parts. The random panning feature can also be useful in adding interest to ear-candy sounds. It's such a simple concept, but it's a great one: I have the feeling that this is one of those plug-ins that will see a lot use, particularly in loop-based music. *Paul White*

$ $53 (discounted to $10 when going to press).
W https://unitedplugins.com

FilterFreak

Get Your Freak On

FilterFreak can do it all – from classic funky envelope filters, massive dance floor sweeps, to pumping up the low end, and classic LFOs or tempo-locked programmable rhythms. Choose from one or two bands of variable shape resonant filters and let FilterFreak deliver warm, fat, analog tones along with surprisingly deep features you can't get from hardware.

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– Tom Elmhirst (Adele, Amy Winehouse, The Black Keys, Florence & The Machine)

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PLUG-IN PARALYSIS

Problems & Prescriptions

We've never had so much choice when it comes to the tools we use to make music — but it's important that you can choose quickly, and with confidence.

ELI KRANTZBERG

Music production was once a simpler affair. Sure, as technology advanced, we started to develop more sophisticated techniques, running ever more mics through increasingly large mixing consoles and out to a tape machine. But even once we'd thrown EQ, compression, reverb and effects into the mix, the production process as a whole remained relatively quick and straightforward. Today, computers, DAW software and, increasingly, plug-ins, lie at the heart of our projects, and these powerful systems give us affordable access to pretty much any 'gear' we want. Plug-ins can be used for so many tasks, and they're almost instantly available, which makes them convenient. The possibilities are endless — and that's all good, right?

Overchoice & Shiny Objects

The thing is, it's precisely because plug-ins are so quick and easy to acquire that they bring with them a significant risk of temptation. We can then end up acquiring so many of them that we can become paralysed by the sheer number of options. Lots of us burn precious time and energy worrying about which compressor or EQ plug-in might be the optimum choice to shape a given sound. And that distracts us from focusing on what's really important: the music, the groove, the feel of a production, and finishing that production!

Psychologically, this can be a tough circle to break: it's well documented that too much choice, or 'overchoice', can lead to a state of inaction. Overchoice is a term first coined by Alvin Toffler in his 1970 book *Future Shock*, and it occurs when someone is presented with so many options that they cannot easily choose between them. Importantly, not only is their ability to make a good decision reduced by this overload of choices, but so too is their satisfaction with their final decision.

I'm not suggesting that choice is inherently bad, though, and studies have shown that having zero choice also results in very low satisfaction — increasing the available choices does initially lead to greater satisfaction, but eventually, as the number of choices grows, our satisfaction level will drop off. We feel more pressure, confusion and potential dissatisfaction with our choices (maybe that other vintage Neve EQ emulation might sound better than the one I spent the last 15 minutes tweaking?). The lesson from all this is that larger choice sets might initially seem appealing, whereas smaller choice sets lead to increased satisfaction and reduced regret.

So let's consider how this applies to DAW-based music production. If we acquire an assortment of plug-ins, we have to choose which to use. On the plus side, there's variety: we know that we almost certainly have tools at our disposal that can help us achieve what we want to achieve. But there's also complexity, which is likely to cause either delay to or dissatisfaction with our plug-in choice. Minimising delay is important — when in the creative mindset, whether writing, arranging or mixing, our best work is done in the immediacy of the moment, often quite intuitively. Trying out six different delay plug-ins in the middle of a live project could well yield worse results in the long run than simply sticking with the first choice and getting the best out of it.

Even if you act on that knowledge, though, and perhaps start weeding your plug-in folder to reduce the overwhelming number of choices, in time it's easy to drift back to a state of too much choice because of something popularly known as Shiny Object Syndrome: "a continual state of distraction brought on by an ongoing belief that there is something new worth pursuing." As with any commercial company marketing products, plug-in developers turn to a variety of psychological and sales techniques to drive us into making purchases.

I'm not saying that ads and marketing are inherently bad. The claims may well have merit, and if there are plug-ins out there that might genuinely make your life

easier or help you achieve better results, you want to know about them, right? But if we allow ourselves to be seduced by the marketing campaigns, acquiring plug-ins can become compulsive, and that can create real problems for us. As well as the problem of overchoice, you can end up spending too much time shopping for or playing with new plug-ins, and not investing the time that's necessary to develop the deeper skills by learning every detail of your tools — you become a jack of all plug-ins, master of none. Consequently, your productivity tanks and you never seem to finish things.

Getting To Know You

How can we overcome these roadblocks to productivity? First, you need to understand what sort of musician/engineer/producer you are, as that will give you a good idea of what sort of tools will likely work best for you. In turn, that should help you focus on which plug-ins might be good acquisitions, and which probably won't improve on what you have.

There are, broadly speaking, three types of plug-in. First, there are modelling plug-ins that emulate the functions of real hardware, along with its sonic characteristics (saturation, phase shift and so on). Second, there are what I think of as 'digital precision' plug-ins, that include useful features like spectrum analysers, detailed metering and, most importantly, comprehensive control that allows you to dive deep into the details in order to achieve precisely the outcome you want. Finally, there are 'assisted' or 'automated' plug-ins. These usually involve an outcome-driven workflow: the user's attention is focused on sonic outcomes, while the complex parameters are adjusted behind the scenes to achieve them. I'd put many of the newest 'AI' plug-ins in that category, obviously, but also 'macro' plug-ins like Waves' One Knob range.

If you're detail-oriented and obsess over minute details, a one-knob effect plug-in probably won't cut it for you. Conversely, if you're a beginner or don't have lots of time/money, a complex Swiss Army knife EQ with tons of features might not be the best choice — maybe try an EQ with a simple interface such as a graphic EQ, where you just have to move a few sliders. Or perhaps an 'assisted' plug-in that listens to and analyses your audio, and generates an EQ curve as a starting point to speed you along. Alternatively, maybe you're a hobbyist who gets a buzz from recreating the vintage studio experience with iconic hardware mixing boards. A console channel-strip EQ emulation with a beautiful skeuomorphic GUI could well be a good choice. In short, they will all get the job done: the question is, which one suits you?

What style of plug-in suits you best — a simple one-slider GUI, one with more to see on the interface and plenty of parameters to tweak, or perhaps one that keeps the details out of sight until you need them?

Try Before You... Bias?

Our bodies are hardwired to crave dopamine, the 'feelgood' chemical that rewards new experiences and new ideas: we get a hit of it every time we obtain a new plug-in. We need to get past that if we want to stop hoarding plug-ins, so how should we make our purchasing decisions?

There are many reasons why people are loyal to the choices they make, and they may or may not be relevant to how you will experience the quality or enjoyment of any given plug-in, so be aware of this when trawling through social media and watching demo videos! The fundamental attribution error when absorbing comments about plug-ins in these places is the assumption that people offer objective, impartial opinions. In reality, we're all prone to cognitive bias. If you need opinions, it's much better to seek out feedback and guidance from a trusted colleague whose opinion you value, and who has experience with the products involved.

For example, if you're in a Slate discussion group and ask about a certain Slate compressor, someone who owns a competing Plugin Alliance product may weigh in to suggest that the PA equivalent is better. Their intervention is almost certainly driven in part by cognitive dissonance: they've invested money in the PA plug-in, and to accept (or even examine!) the possibility that the Slate one might be as good/better forces them to reassess their decision. It could even lead them to question their understanding of compressors! It's much easier to confirm their pre-existing belief that the PA compressor is best. »

» Another cognitive bias that drives behaviour is the ‘halo effect’, whereby our impression of one aspect of something influences our perception of other aspects. For example, if a plug-in has an attractive GUI, we might be mentally prepared for it to sound as good as it looks. We might experience the halo effect because focusing on a certain aspect when forming an impression about a plug-in reduces our cognitive load — the more factors we consider and the more complex they are, the harder it is to form a valid impression. So if a developer’s marketing triggers a dopamine hit by focusing on one specific aspect of the plug-in, we may neglect the bigger picture — we won’t ask ourselves if this plug-in will serve any real need beyond the dopamine hit.

It’s also easy to succumb to ‘FOMO’ (fear of missing out) during time-limited sales. The reality is that the sales always come back sooner or later, and in any case they’re usually long enough (at least 48 hours) that you can download and try a demo — doing this has saved me from several purchases I’d have regretted.

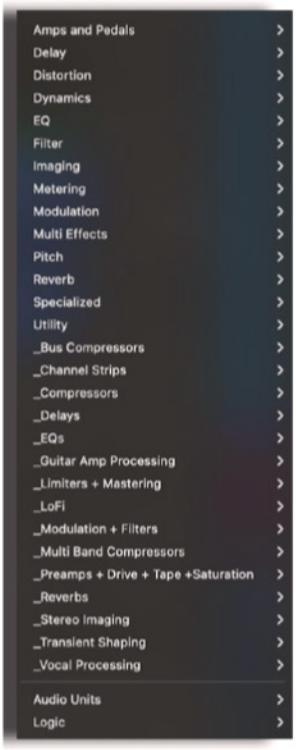
To recap, try and assess the value of adding any plug-in to your collection. Ask what value it brings to you, and what sets it apart from the competition. Does it offer unique features and functionality? A better quality of sound? Ease of use?

Perfection Versus Productivity

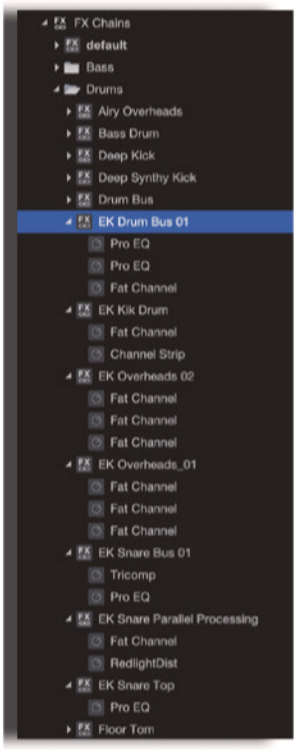
Ever heard the phrase, ‘The perfect is the enemy of the good’? This lies behind a decision-making strategy known as ‘satisficing’. The idea is that always aiming for the optimum solution consumes time, energy and resources you need for things that *must* be perfect. Aim to complete every task to a satisfactory standard rather than ‘perfection’, and you’ll be more productive. The best way to get into this mindset is to restrict your choices: limit what your eyes can see in your plug-in folder!

Most DAWs let you organise at least third-party plug-ins into custom folders, and doing that makes choosing more efficient. When looking for a reverb, for instance, it helps if they’re all in one folder, so you don’t have to recall a plug-in name or which developer’s reverbs you’ve installed. Studies show that our brains process options most efficiently when there are seven or fewer of them. So if you must have 20 different reverbs, consider subfolders (eg. algorithmic reverbs, convolution reverbs and special effects), each with seven plug-ins or fewer. This is also an opportunity to do some weeding: if you’ve not used that free distortion plug-in you downloaded a year ago, do you really need to be staring at it when looking for a distortion?

Yet, no matter how well your plug-in folders are organised, it’s easy to forget what you have, and what worked or didn’t previously. To help with that, why not use your DAW’s preset system to store and recall chains of effects with specific settings? Maybe in one mix, you created a killer parallel drum-processing setup, with a saturator, an EQ, a transient shaper and room tone, each a different plug-in. This chain may or may not work in future mixes, or perhaps part of the chain will work with tweaks. Save it as a preset and that’s easy to figure out! Having variations of these chains can help combat overchoice fatigue too. You can have too many, of course, but you’re spending less effort choosing and tweaking plug-ins, and even if they don’t work in the new context, they may jump-start an idea, or point you in an interesting direction. The point is to help you make a choice quickly and move on.



Stock DAW plug-ins often can’t be moved in your plug-in folder, but you can usually organise third-party plug-ins into meaningful folders and subfolders for easier, less overwhelming access.



Some customised effects chains, ready and waiting for drum processing in Studio One. Such chains can not only save time, but reduce the burden of decision making — so you can keep your focus on making music.

Here’s one of my DAW templates. It combines active, offline and bypassed plug-ins, which helps to limit my choices and facilitate speedy experimentation and decision making.

Templates With Disabled Options

An underrated DAW feature is the ability to load project templates with tracks or plug-ins disabled. If you like having lots of choices immediately available, this is just the ticket. You could have, say, your 10 drum parallel-processing busses ready and waiting to be auditioned at the click of the mouse, with no need to hunt through plug-in or effects-chain menus. This can also facilitate easy creative experimentation, because

you can use multiple choices. By the way, this applies to instrument plug-ins just as much as processing: why not have your five favourite Rhodes patches ready to go for easy auditioning, or a whole orchestra ready to call on but not consuming computer resources?

Of course, templates with an excessive number of unused instruments and effects plug-ins just 'standing by' can hinder as much as they help. If, for example, on loading there's a problem with a plug-in that needs updating, it can be time-consuming to troubleshoot. And even though the load time isn't increased, with all the additional disabled tracks the file size can swell — with each 'save', your DAW is doing heavier lifting than necessary, and each 'save as' can duplicate content that's irrelevant to the project. With all the tracks, hunting down routing problems that may arise can also become more cumbersome.

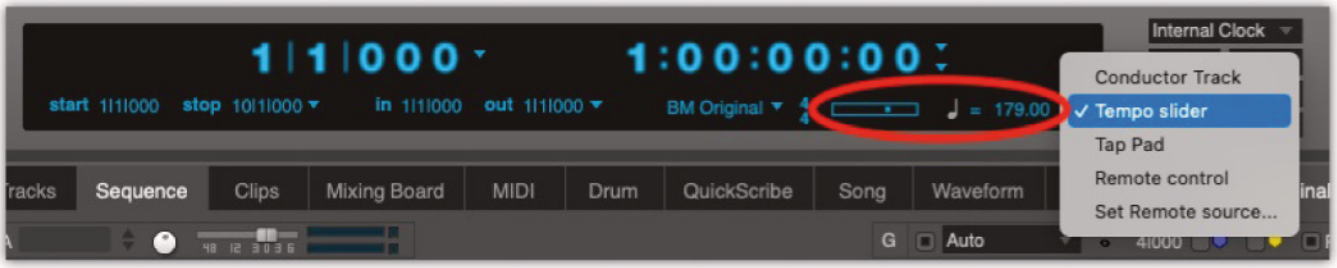
So I prefer a hybrid approach, combining the advantages of templates with the benefits of stored effects chains. For instance, I have a multitrack drum-recording setup in my template. The tracks are all set with the proper inputs, named and colour-coded, and a few of my favourite third-party plug-ins on each track as a starting point. I also have a few of my favourite instruments ready and available for quick inspiration. I have sends for two independent headphone mixes and a comfort reverb in place on each track. But beyond these basics, I prefer to call up channel strips I've saved over the years for things like guitar or vocal processing. Remember the rule of seven? This is the same: I don't want my template to visually overload me.

Cognitive Nudges

It all sounds so simple doesn't it? But denying yourself a dopamine hit can require the self-discipline of a monk! So, for when you're next hovering your cursor over that 'buy now' button, I'll leave you with a few cognitive nudges that may help:

- More is not better. More choice often leads to worse results, if you factor in the indecision and delay it can introduce in the heat of the creative moment.
- While sometimes new is better, often it's not — the more important question is, will it be better *for you*?
- Remind yourself that not every plug-in will lead to better mixes. (Your specific choice of Pultec emulation is probably not what's preventing you from achieving a great kick sound.)
- Set an arbitrary limit to the number of new plug-ins you add to your system. Even if they're free downloads, resist!
- Make a mental note of how often you use the plug-ins you have. When that dopamine buzz is pending, this can help jolt you back to reality.
- Remove temptation from your inbox. If dieting, you might change your route so you don't walk by the bakery. Similarly, you can unsubscribe from plug-in reseller mailing lists.
- Make the effort to see past the hype. How often do true 'gamechangers' come along after all?
- You're not FOMO-proof! Ask yourself how much of what is driving your desire the fear of missing out on something new, sexy, and cool; versus something you really need and will benefit from. Try 'wait and see': even if it's in a time-limited sale, avoid buying on impulse; sleep on it.
- Research every new plug-in you acquire to understand its potential. Walkthrough videos might be better than review videos, as they remove some of the subjectivity, making it more likely you'll use your own judgement.

While things might never again be as simple as the 'point and record' beginnings of our craft, with some careful management, you can minimise distraction and stay focused on what is most important: making great music!



Screen 1: If the imported audio has a constant tempo, try simply adjusting the tempo slider (with DP's click enabled) to align the sequence tempo with the audio.

Need to turn your audio and MIDI recordings into usable sheet music? DP has you covered.

MATT LAPOINT

Transcribing is the process of creating readable music notation from a music recording or performance — or, in a DAW context, from MIDI data. Melody, harmony and rhythm are the basic musical elements that are transcribed into individual instrument parts. You can use the resulting transcription to create scores or lead sheets, but it can also be used to employ interesting production techniques, such as doubling parts. Digital Performer includes a powerful tool set for transcribing both MIDI and audio recordings.

Taking Notes

Internally, DP is always transcribing MIDI notes into notation, and has a powerful engine for doing so. The extensive QuickScribe notation editor displays the notes 'on the page' and provides many settings for customising staffs, ledger lines, measure spacing and much more. Tool palettes provide controls for text, chord symbols, articulations, and virtually all commonly used musical symbols. However, an essential requirement for a clean and accurate transcription is the relationship of the notes to the sequence timeline. If the audio or MIDI was recorded 'freestyle' and the notes don't align with the measures and beats in the Conductor track, the transcription will look more like a train wreck than readable music notation. An accurate tempo map that aligns with the music is essential to achieve the best possible transcription results.

Transcribing Audio Recordings

Most of the time, though, when you are faced with the task of transcribing music, your source material is an existing audio recording. So, let's examine a few techniques for creating a tempo map that aligns with previously recorded music that you import into DP.

First, try to obtain the highest quality audio available. Uncompressed formats like WAV and AIFF are preferred. After the tempo is adjusted, we'll discuss slowing down the audio later in this article. The higher the audio resolution, the slower the audio can be made on playback (as discussed later).

Import the WAV or AIFF into DP using File / Import Audio, or by dragging the file from your file browser into the left side of the Tracks window. DP will automatically create a new track matching the channel format of the imported file. Files can also be dragged into an existing audio track or the Soundbites window (and then added to audio tracks manually).

Finding The Tempo

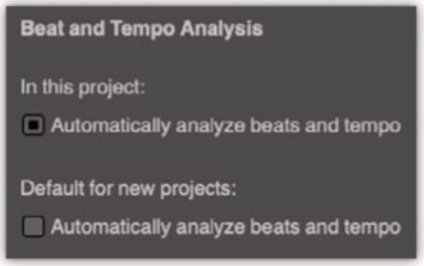
Once the audio is in a track, edge-edit the left edge of the Soundbite to the initial downbeat of the music, eliminating any dead space at the beginning. If there are pick-up notes before the first downbeat, edge-edit to the start of the first pick-up note. Place the soundbite at bar one of the timeline (the very beginning of the sequence). If there are pick-up notes, place the soundbite in bar one such that the initial downbeat of the music lands on the downbeat of bar two.

Once placed, here are a few ways to align the sequence tempo with the audio tempo.

Tempo Slider: If the source music has a constant tempo, try simply adjusting the tempo slider while listening to the music with DP's click enabled (see Screen 1). Double-click on the metronome icon in the control panel to choose a click sound, level, accent, and output. Often it doesn't take long to adjust DP's tempo to be locked in with the music.

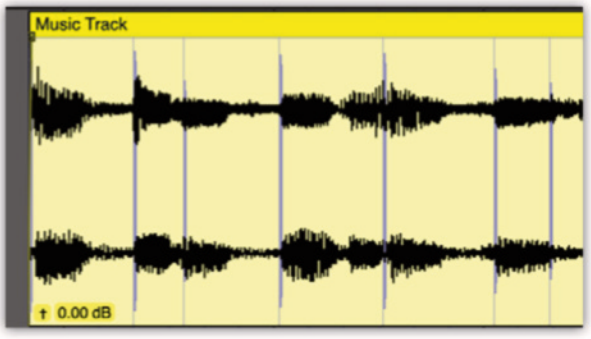
Tap Tempo: You can also try to match the tempo of audio with a constant tempo using DP's Tap Tempo feature. By default, this feature requires clicking on the on-screen Tap Pad in rhythm to enter the tempo. If, instead, you wish to tap on a computer key, MIDI keyboard key or a MIDI controller pad, search for Tap to Enter Tempo in the Commands window to assign your desired key or pad to tap tempo. Tap along until DP's click matches the audio.

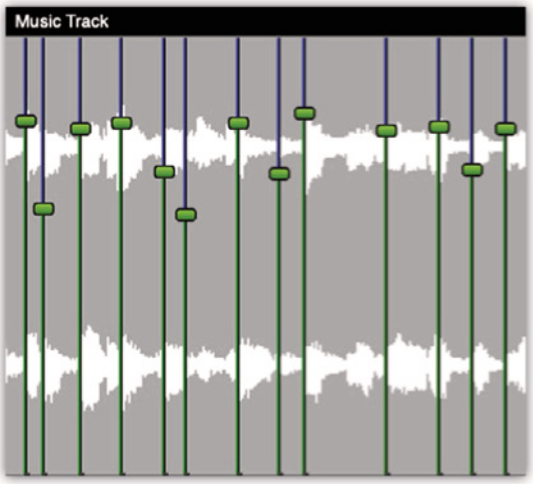
Audio Beats: DP has a useful global setting for beat and tempo analysis. Under DP / Preferences / Background Processing, enable beat and tempo analysis for the current project (see Screen 2). When enabled, DP quickly calculates beats and tempo information for soundbites in the background. This is



Screen 2: With these preferences enabled, DP will automatically find beats and tempo in audio files.

Screen 3: Detected beats are indicated by vertical blue lines.





Screen 4: You can manually edit beats in the Waveform editor to fine-tune the beat map. Use the fader handle to indicate the strength of the beat.

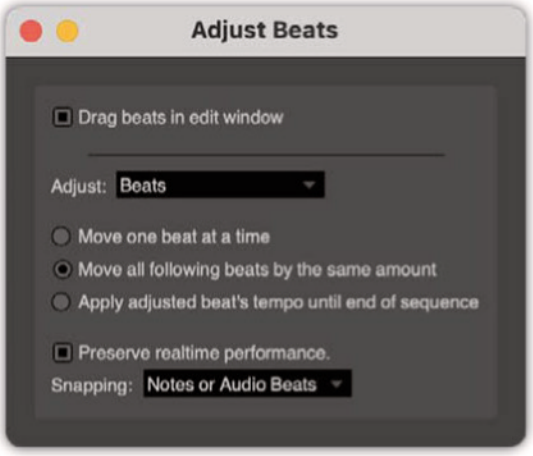
true for both imported audio files and newly recorded audio tracks.

DP bases audio file tempo on the analysed beats. In fact, the tempo can't be calculated at all unless DP finds the beats in advance. The beats in soundbites are represented by blue lines that indicate attacks and transients in the audio (see Screen 3).

There are two ways to alter the detected beat sensitivity. Select the soundbite and choose Audio / Audio Beats / Find Beats. Drag the slider right or left to increase or decrease the sensitivity. DP will quickly re-render and display the results, adding or removing blue lines according to the sensitivity setting. Hit Apply to keep the results. If Find Beats is still unable to catch all of the necessary beats, view the soundbite in the Waveform editor. One quick way to do this is to right-click on the soundbite and choose Open in Waveform Editor. At the bottom of the window, choose Beats under the Edit Mode menu. This will display green lines indicating the found beats. Use the pencil tool to click and add additional beats as needed (see Screen 4). Selected beats can be deleted with the delete key or muted with the mute tool.

If the beats have been modified, re-analyse the soundbite tempo: select the soundbite and choose Audio / Soundbite tempo / Analyze Soundbite Tempo. To map the soundbite's new tempo map to DP's Conductor track, make sure DP is set to Conductor track tempo mode, and then select the soundbite and choose Audio / Soundbite tempo / Adjust Sequence to Soundbite tempo. DP will create a new tempo map that matches the soundbite tempo. If the audio beats and tempo were accurately calculated, the music should now be locked to DP's click (and timeline).

Record Beats: Record Beats is similar to Tap Tempo: you tap along with audio playback using a key on your MIDI controller to record your taps as tempo events in the Conductor track. The resulting tempo map should then align with the audio. Record Beats can also be used for a MIDI performance that was not recorded to a click. The DP User Guide PDF is a good resource for learning more about these features. MOTU's YouTube channel, MOTU TV, has a specific how-to video about Tap Tempo.



Screen 5: Use Adjust Beats to drag beats and bar lines in the time ruler to align them with beats and bar lines in the audio.

Adjust Beats: DP's Adjust Beats feature is another way to manually make local tempo adjustments around MIDI notes, markers and audio beats. Put tempo control into Conductor track mode and view the soundbite in the Sequence Editor. Open Adjust Beats from Project / Conductor Track / Adjust Beats (see Screen 5).

This feature facilitates anchoring notes, audio beats or markers to timeline measures, beats, or fixed durations. It can preserve the performance and simply change the tempo to align the timeline to the audio or MIDI material. It provides options for moving single beats, moving all following beats by the same amount, and applying the adjusted tempo until the end of the sequence. With the Adjust Beats window open, click on the audio waveform in line with the time grid for the chosen beat value. A shadow will appear and the pointer arrow will change to a different shape. Drag left or right until a red line appears (Screen 6).

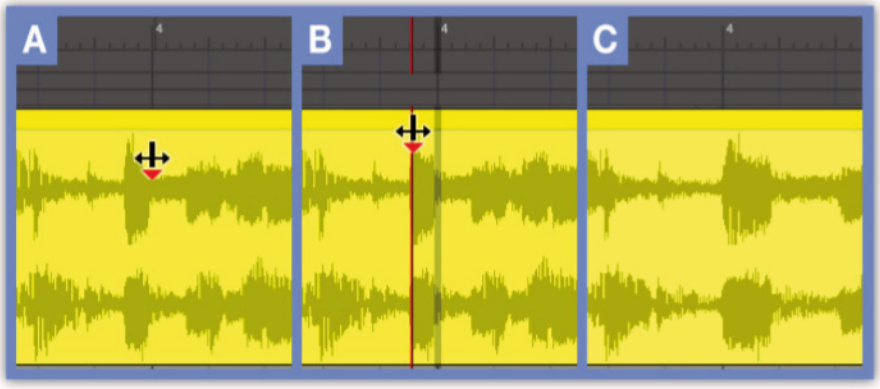
Based on the Snapping settings in the Adjust Beats window, drag and snap the red line to the note or audio beats desired. When you let go, DP updates the Conductor track to align the note or audio beat to the grid. It's best to work left to right while realigning beats. Again, the DP manual and MOTU YouTube channel are good resources for more information on using Adjust Beats.

Ready For Transcription

Now that the audio is aligned with DP's timeline, you are ready to use whichever of the transcription methods described below that best suits the material.

One of the easiest and most direct ways to transcribe audio is to slow down the track and play along with the audio »

Screen 6: In this example, the Adjust Beats cursor is being used to grab the downbeat of measure 4 (A), which is then dragged to the beat transient to the left (B). The result is that beat 4 now aligns with the beat (C). The audio didn't change. Instead, the timeline did.



» material, recording your MIDI performance. Fortunately, MOTU's implementation of time-stretching doesn't change the audio pitch when slowing down monophonic or even polyphonic soundbites. Depending on the complexity of the music, it can be incredibly helpful to slow down the tempo to listen more critically, pick out specific musical parts, and better discern notes and phrasing. Because the music has been aligned with DP's timeline, your newly recorded MIDI material will transcribe properly, even at considerably slower tempos, to mirror melodies, harmonies and rhythms from the audio guide.

First, turn on the Stretch setting for the track from its Track Settings menu in the Sequence Editor. The audio track will now follow both tempo slider changes and Conductor track tempo changes immediately, on the fly, without any rendering.

Next, to ensure the best possible audio quality, go to DP Settings / Pitch and Stretch, and set ZTX quality in the current project to Standard Best from the Pitch and Stretch Playback menu.

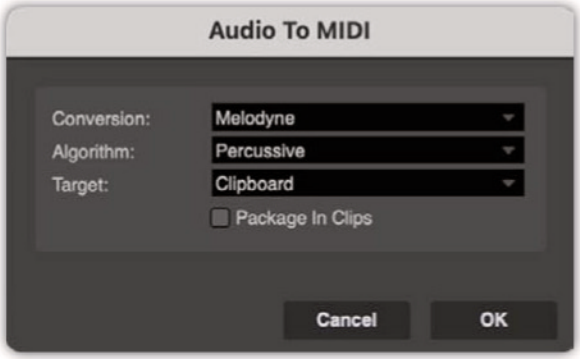
You are now ready to slow down the tempo. If the conductor mode is set to Tempo Slider, simply type in a slower tempo for playback. If the mode is set to Conductor track, select the tempo data in the Conductor track and use Region / Scale Tempos to slow down the tempos by subtracting or scaling values. Feel free to experiment with fairly extreme values. Remember, the audio quality doesn't need to be perfect, just good enough for you to be able to play along.

Once the MIDI has been recorded or pencilled in, the tempo can be brought back to the original speed.

Alternately, you can apply offline processing, which offers even greater resolution but requires the use of the Scale Time window from the Region menu. This reprocesses the audio and can take time to process based on the chosen quality settings and the duration of the soundbite.

Audio To MIDI

DP11.2 (and above) provides state-of-the-art audio-to-MIDI conversion in three forms: single melodic lines, rhythms, and polyphonic note detection. These are not only extremely helpful when transcribing parts in QuickScribe, they are also useful for extracting parts for producing new tracks or parts inside DP. In the Sequence editor, it's best to make a time-range selection of the soundbite for the audio-to-MIDI conversion. Control-drag to select, or use the I-Beam tool to do so. After the soundbite is selected, go to the Edit menu and choose Copy Audio to MIDI. The Audio To MIDI window provides three conversions in the form of Melodyne, PureDSP, and beats (see Screen 7).



Screen 7: Use Audio to MIDI to quickly convert audio tracks to MIDI data.

Melodyne uses the Melodyne engine from Celemony. PureDSP uses MOTU's own pitch algorithm and is designed for monophonic, single-note tracks only. Beats uses DP's internal beat analysis and puts the transcribed MIDI notes on one pitch. When Melodyne is selected as the conversion algorithm, the Algorithm menu provides seven options:

- **Automatic:** Performs a pre-pass to figure out the best algorithm to use based on the material.
- **Universal:** Best for time-stretching and transposing where individual notes are not needed.
- **Percussive:** Best for conventional drum and percussion sounds or loops.
- **Percussive Pitched:** Best for 808 kicks and toms, tabla, and similar percussive sounds with a pitched component.
- **Melodic:** Best for single lines like solo voice, saxophone, flute, and monophonic bass.
- **Polyphonic Sustain:** Best for longer polyphonic sustain notes (such as legato strings).
- **Polyphonic Decay:** Best for shorter notes like pizzicato strings and plucked instruments.

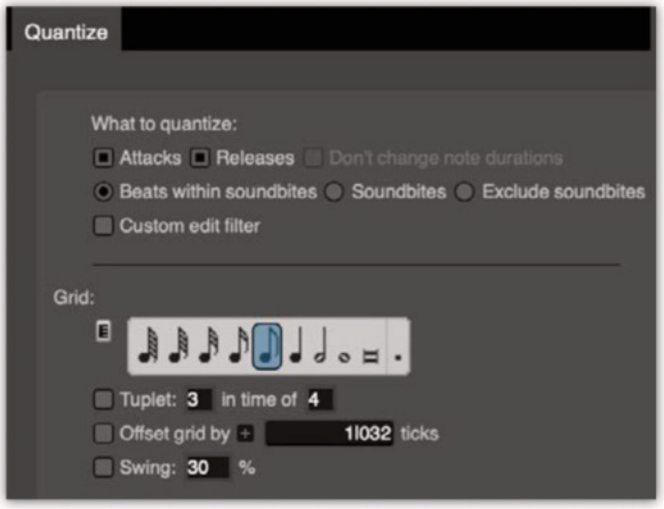
When the conversion finishes, the Target window provides four options for how to use the results. The Clipboard option copies the data to the DP clipboard so it's ready to paste into a MIDI or VI track. Alternatively, you can place the material into an existing Clipping window. Or, you can create either a a new unassigned MIDI track or instrument track, populated with the converted MIDI data. The Package As Clips tick box provides the option to create the MIDI data in the form of a Clip.

Quantising The Results

After converting audio to MIDI, it might be useful to quantise the notes to achieve the best transcription results (see Screen 8).

Depending on the accuracy of the conversion, try quantising the Note On and Note Off messages to fix the timing and eliminate errant note lengths. The Smart Quantize window (Region menu) is designed to simply tighten attacks and durations for a more accurate note display in DP's QuickScribe window. It's a good idea to do a Save As or duplicate track takes when performing destructive processes like Quantize and Transpose.

Screen 8: For best results, quantise MIDI data that has been converted from audio.



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Timeline and Edit selections are the key to editing in Pro Tools.

JULIAN RODGERS

This month I'm going to look at selections in Pro Tools. We often take selections for granted, but as they are so fundamental to operating a DAW, they credit a closer look. They do after all define exactly which material we are targeting for any particular action or edit. This is a big area, but I'm going to cover a few fundamentals and then highlight a few less well-known techniques. Hopefully there will be something new here for most people.

Something I have to cover first, because it is so central to understanding how Pro Tools works, and because it's the source of much confusion for people who don't understand it, is the two types of selection in Pro Tools: Timeline and Edit selections.

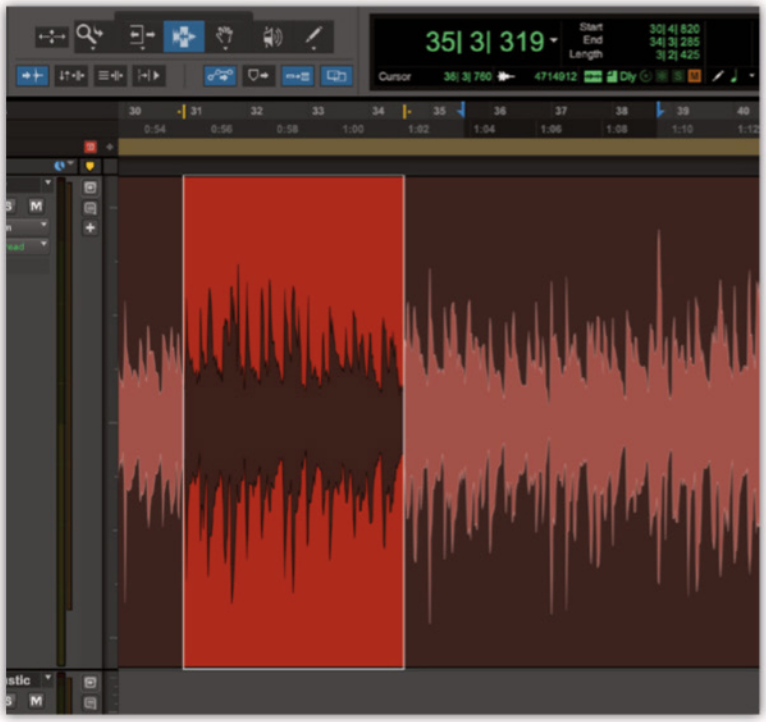
Time & Place

A Timeline selection defines what gets played back when you hit the space bar. It can be any duration, from zero to the whole length of the session. Its length is denoted by a pair of half arrows in the timebase ruler, which are either blue or red depending on the record status of your transport. If the length is zero, you'll see what looks like a single downward-facing arrow, but this is actually two markers next to each other. If the playback selection has zero length, Pro Tools will play from that point until you stop it. If the selection has a length, it will play just the timeline selection and stop, unless the transport is set to loop.

An Edit selection is the area of the timeline that will be affected by an edit, which could mean anything from copying, deleting or moving to processing using AudioSuite or Clip Gain. An Edit selection can include partial and whole clips, a group of clips, or an area covering any amount of the timeline across a single track or multiple tracks. Like a Timeline selection, it can also have zero length, in which case it is represented by a flashing line on the timeline called the Insertion Point. If the Edit selection has a duration (ie. the in and out points aren't in the same place) it will be represented by a box of lighter or darker colour, depending on whether you are using the Dark or Classic UI Theme. The in and out points of Edit selections are represented by orange markers in the timeline ruler. However, in normal use, these are usually obscured by the blue Timeline selection markers, for reasons we'll discuss next.

The Link

The distinction between these two types of selection is usually invisible to users as, by default, the two selection types are linked. In other words, if you make an Edit selection and hit Play, the Timeline selection corresponding to that Edit selection will be played back. This is really useful, but it can be desirable to unlink them in some situations, and if you've never unlinked them before, the results can be confusing. If what you select doesn't get played back, you need to press Shift+/ to relink them.



Edit selections will show up as either darker or lighter patches of audio, depending on whether you're using the Dark or Classic theme. Their in and out points are represented by orange lines in the timeline, whereas the in and points of Timeline selections are shown in blue.

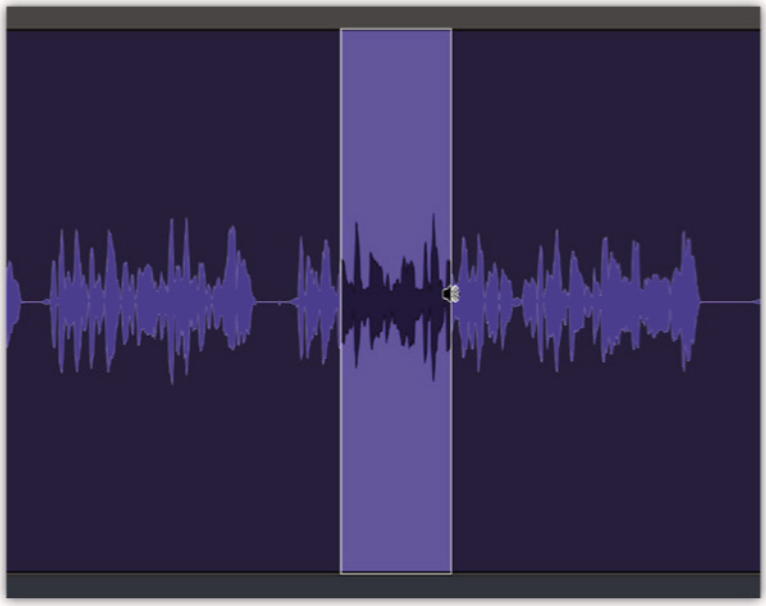
Making selections is pretty straightforward: there's a dedicated tool for it, unsurprisingly called the Selector tool, and if you click and drag you can select an area of the timeline. You can get more precise, though. Once it's created, you can finesse your Edit selection by adding Shift and moving the in or out point. If your audio waveform is transient-rich and the beginning of each note or hit is easy to see, then the inordinately handy Tab to Transients feature can be used. Normally, if you hit Tab, the Insertion Point jumps to the end of the clip or the start of the next one. Hit Cmd+Opt+Tab (Ctrl+Alt+Tab on PC) to turn on Tab to Transients, and it will jump to the next transient instead. Use Shift with Tab to create Edit selections based on transients (or MIDI notes).

But what if your material isn't transient-rich? What if it's a sustained sound that changes notes smoothly? The simplest way to quickly select the material you want is to use your ears. If you can hear where one note ends and another begins you can create selections on the fly using the arrow keys. Hit the down arrow to create the in point of your Edit selection and the up arrow to create the out point, and if you need to finesse these, you can use the Shift key as mentioned earlier.

The Audition feature of Pro Tools is useful for checking your in and out points. Use the 6, 7, 8 and 9 keys on the alphanumeric keyboard to preview the audio up to your in point, from your in point, to your out point, and from your out point, respectively. These Audition keys are incredibly useful when tracking too. You can use them to let your artist know exactly where you're recording from and to, and to audition the pre-roll.

Scrubbing Up

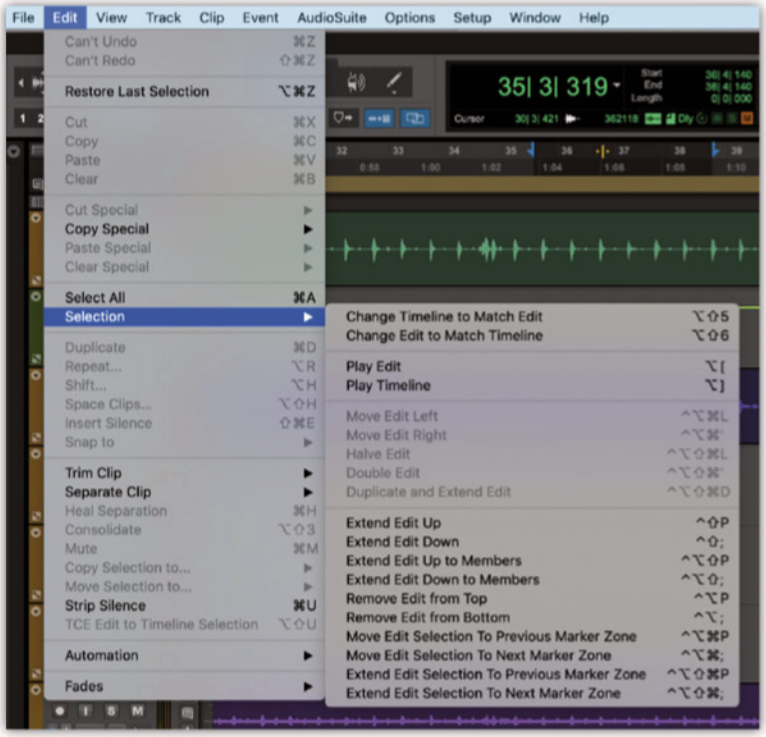
There is another way to create Edit selections by ear: using the Scrub tool. This tool, which looks like a loudspeaker, can be dragged right or left to simulate the effect of rocking tape backwards and forwards across the tape head, which is the way people used to find edit points in the tape days. Although useful for amusing



The Scrub tool lets you make your selections by ear. Usefully, the Scrub tool icon disappears while scrubbing so as not to obscure the waveform.

record scratch effects, it really comes into its own for precisely locating hard-to-find edit points, and can be used to create Edit selections just as easily. As long as Edit Insertion Follows Scrub/Shuttle is enabled in the Operation tab of the Settings window, you can use the Scrub tool to find the appropriate in point, and hold Shift to scrub to the desired end point.

A very useful feature that has been around since Pro Tools 8, but which some users seem not to know about, is Dynamic Transport. This feature, enabled from the Options menu or by pressing Cmd+Ctrl+P (Ctrl+Start+P on PC) allows the user to drag the Timeline selection up and down the timeline, making auditioning looped sections and moving between sections quick and easy. In Dynamic Transport mode, the Edit and Timeline selections are linked, but if you want to move an unlinked Edit selection without changing its length, you can do this by Option-dragging either the in or out point in the ruler to slide the selection up and down the timeline.



The Edit / Selection menu offers a host of options for changing your selections.

There is a whole collection of selection actions under the Edit menu. Two that are really useful are Move Edit Left or Right, which move the selection along the timeline without changing its length. Two more favourites from this menu are Halve and Double Selection Length. The keystrokes for these are listed in the menu and are very worth getting under your fingers if this is something you think you'll use regularly.

Ups & Downs

The ability to move selections up and down vertically from track to track is likewise very handy — and, returning to my explanation of Edit selections at the beginning of this article, the flashing Insertion Point can be thought of as an Edit selection with zero length, and the ability to move this up and down from track to track is also incredibly useful. If you're moving a clip vertically from track to track, you can constrain it so it won't move from side to side while being moved by holding the Ctrl key (Start on PC). But what if, if for example, you're moving a bass note so it falls precisely on the kick drum in a session with no timing grid? In that case, using the P key for up and the ; key for down, you can move the insertion point from the kick drum hit (easily located using Tab to Transients) to the bass track. Then use the equally handy trick of Ctrl-clicking on the suitably separated errant bass note with the Grab tool to snap the bass Clip to the correctly positioned Insertion Point, precisely located under the kick drum hit but on the bass track. This is a must-have technique when working on material in Slip mode

Lastly I'll offer a lifesaver tip for making Edit selections. If you accidentally click away from your carefully set up selection and it disappears, hitting Cmd+Z won't help you. Selections don't go to the undo queue. However, they have their own command: Restore Last Selection can be found under the Edit menu and can be invoked using Cmd+Opt+Z (Ctrl+Alt+Z on PC). This is most commonly needed when you have Insertion Follows Playback accidentally selected, but explaining what that is will have to wait until another day!

JOHN WALDEN

Cubase now boasts a number of decent compressor plug-ins. Which should you choose for vocals?

When it comes to modern vocal production, compression is pretty much a ubiquitous part of the processing chain. In most cases, the primary aim will be to manipulate the dynamic range so that, when placed within the final mix, every part/phrase/word of the vocal comes across clearly. Having achieved this, a secondary consideration might be to impart some character or attitude to further enhance the vocal, either tonally, or in terms of finessing the attack or sustain elements of the individual words.

Cubase now offers a number of stock compressors, including two new choices added in v13: Vox Comp (for Pro and Artist users) and Black Valve (Pro only). So how do you decide which of these compressors to use on your vocals? To help you choose, I grabbed a suitable vocal track and put the two 'new for v13' offerings up against the long-standing standard Compressor plug-in, to see if I could identify the role that each might play in achieving these typical compression aims. You'll also find, on the *SOS* website, some audio examples that demonstrate what I'm talking about: https://sosm.ag/cubase-0724.

One Knob Wonder

As can be seen in the main screenshot, Steinberg have taken the keep-it-simple approach to Vox Comp's control set. The documentation suggests that it features a 'highly adaptive' algorithm that's designed for processing vocals. Given that there is a specific option to engage a low-latency 'live' mode, I assume that there's usually a 'look-ahead' stage of a few milliseconds that lets the plug-in see what's coming and adjust the processing to suit.

Once inserted, you simply dial in Vox Comp's Threshold knob to achieve the required amount of compression (there is no user control over the attack/release or ratio settings; presumably, these are built into that adaptive algorithm). If you need to level-match the processed signal to the original, you can adjust the Output knob. I've not used it here, but you can also use Vox Comp for parallel compression without setting up any routing in Cubase, thanks to the Dry/Wet knob.

As is easily demonstrated by the audio examples that accompany this workshop, the results can be very good indeed. My example vocal features noticeable level differences between the verse and chorus sections, and several words within the verse section that are quieter than some others. Using the settings shown within the first screenshot, Vox Comp's one-knob magic does a very good job of levelling out these quite marked differences (the loudness range across the example goes from 13.1 LUFS unprocessed to 7.8 LUFS after processing) while still maintaining a fairly natural sound.

These changes are reflected in the vocal's waveform, as shown in the second screenshot. Sonically, it's impressive stuff: a much more even vocal performance is achieved effortlessly, the words and individual syllables can all be heard clearly, and it still sounds fairly natural. If in doubt about your vocal compression choice, then, Vox Comp should be a very good place to start, regardless of your confidence level and/or experience with compressors featuring fuller control sets.

Added Character

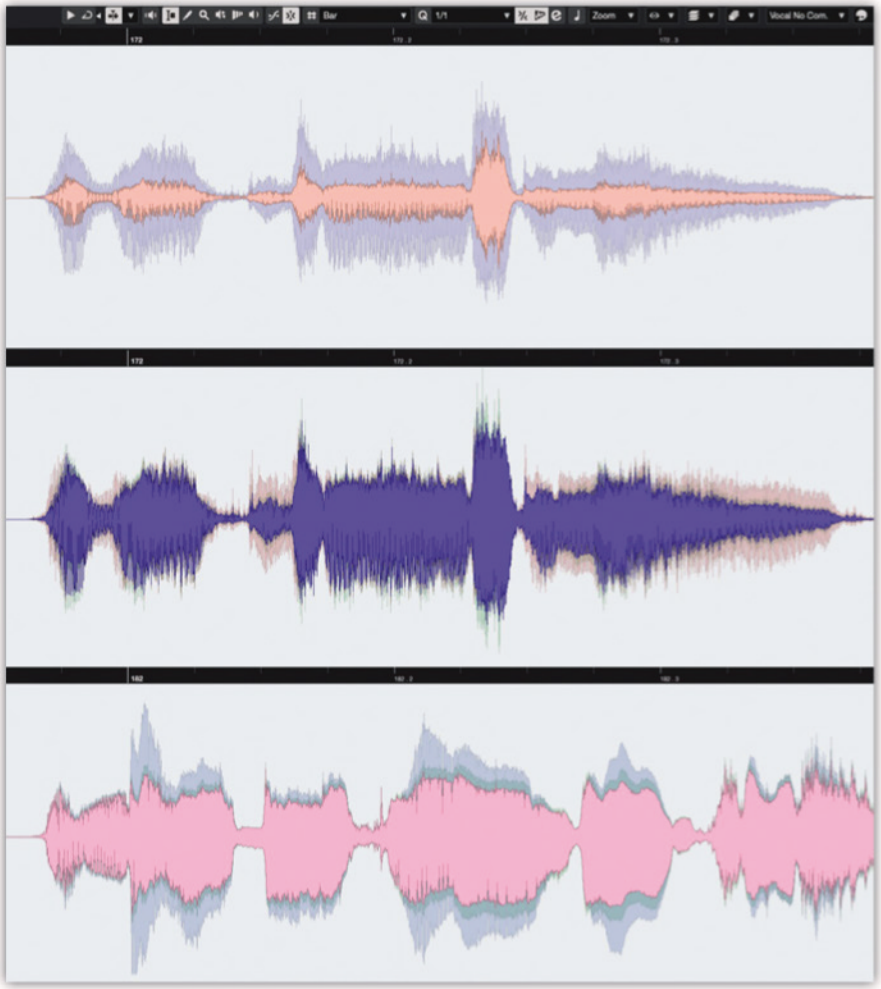
Users of Cubase Pro 13, though, have the option of using the Black Valve plug-in, which emulates a classic tube-style compressor. There are more knobs here (five, compared with Vox Comp's three) and it's not designed solely with vocals in mind, but it still offers ease of use and is certainly useful in this context. The plug-in emulates two important characteristics found in Black Valve's hardware inspirations: a relatively slow attack time and the option to add some preamp-based saturation to the sound.

The Dry/Wet and Output level controls function as they do in Vox Comp and, again, there are no individual attack, release or ratio controls. But as on lots of hardware valve and optical compressors, the Peak Reduction knob acts as a sort of 'macro' control, influencing both threshold and ratio parameters, with higher values providing more overall compression. The Drive knob controls the signal level going into the modelled tube preamp and, pushed suitably hard, adds some saturation to the sound. The gain control then provides a post-preamp gain-staging option before your signal hits the compressor.

As shown in the screenshot, I set a pretty high Drive level to make the tube saturation sound a little more obvious in the audio examples (it's particularly noticeable on the harder consonants), but note that if you wanted to exaggerate this further then you could apply a few extra dB of gain prior to the signal reaching the plug-in — this would drive the tube stage even harder. I didn't think it was appropriate for this particular vocal part, but the modelled tube does break up in a very pleasing fashion so it's an option to bear in mind in a different context.

The combination of saturation and compression resulted in a shift from the

What's your pick for routine vocal compression duties? Vox Comp, Black Valve or Compressor?



Changes to the vocal waveform as a result of the compression settings shown in the first screenshot. Each panel shows a short example from the same vocal performance — the top and middle panels are from the quieter verse section, while the bottom panel is from a louder chorus section. The top panel compares the unprocessed vocal (pink) with the Vox Comp version (pale blue). The middle panel shows the three different compressors: Vox Comp (blue), Black Valve (pale red) and Compressor (pale green). The bottom panel shows three different configurations for Compressor, using the settings shown in the other screenshot (in pale green), with the same settings but an 8:1 Ratio (in pink), and with slower attack and release settings (in pale blue).

original 13.1 LUFS to 6.2 LUFS, and the much more even nature of the processed performance is clearly audible. Looking at the resulting waveform display, my Black Valve settings produced a slightly greater increase in the loudness of the quieter sections, but more noteworthy is the slightly higher levels of some note onsets. This is not a surprise given the somewhat slower attack time, and although it's a small difference, to my ears it results in the starts of words having a slightly stronger emphasis. That might or might not be a desirable result, depending on the captured performance and the musical context, but it's a useful option to know about. So if you're looking for a compressor that might impart a little additional character to your sound, while still being very easy to use, Black Comp is a great candidate — I'd certainly check it out before choosing to splash the cash on a new third-party compressor plug-in.

Clinical Precision

Cubase's standard compressor, called simply Compressor, is designed for those occasions on which you want precise control over all the elements of the compression process. It requires a little more user input (and probably experience), but the results are clean and controllable. As shown in the screenshots, for the main example used here I used a combination of a modest (4:1) ratio and fast attack and release times. I then just adjusted the threshold to taste until I felt the vocal was sitting at a more consistent level. Analysis subsequently indicated this resulted is a loudness range of 8.0 LUFS.

As shown in the short waveform, the most obvious feature of the Compressor processed version is that the loudest parts of the vocal are not compressed as much as with Vox Comp and Black Valve. This is most likely because the fixed 4:1 ratio is lower than the adaptive ones of Vox Comp and Black Valve are on average — a little experimentation revealed that an 8:1 ratio in Compressor produced something closer to the results of Vox Comp. Keeping the ratio and threshold fixed and varying Compressor's attack and release produced exactly the sorts of changes you might expect in the waveform, making it possible to experiment with the balance between note attacks and note sustains with a level of control that's not really possible with the simplified controls offered by Vox Comp and Black Valve.

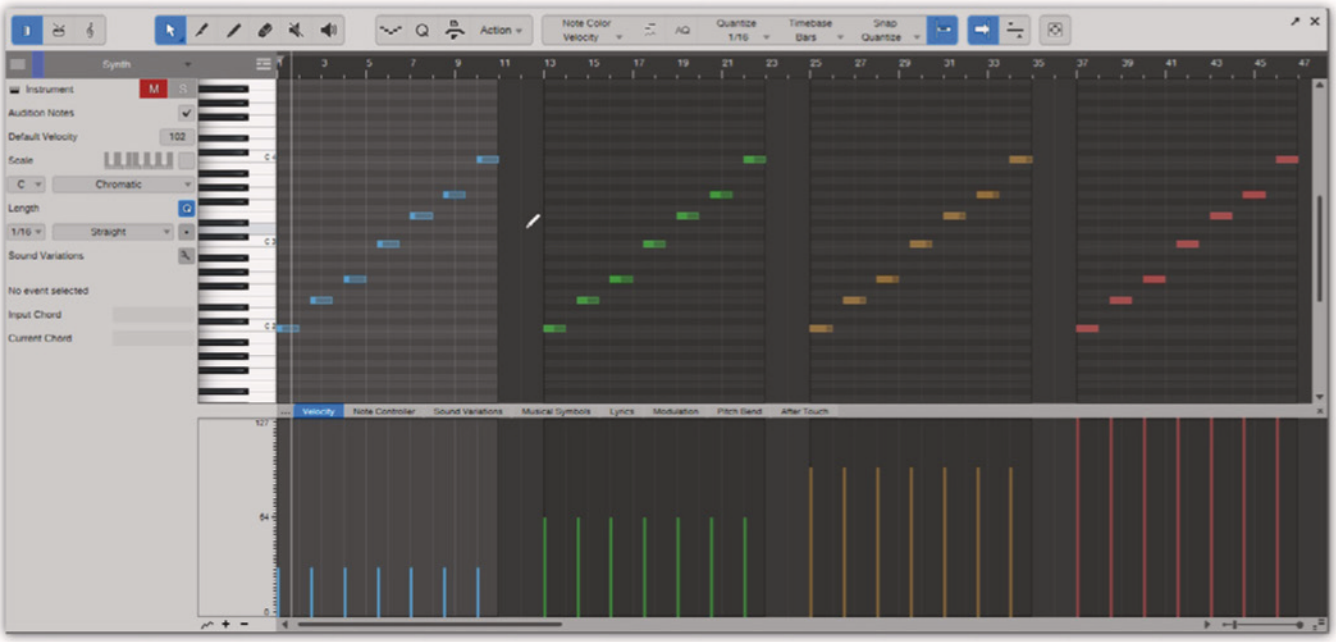
Again, to my ears, the results in this example sound very clean. The overall level of compression achieved in this case was in the same ballpark but, of course, the user has much more precise control over how that's achieved. If you're more of a control freak, Compressor is a great choice — and note that if you want a less clinical-sounding result, you could always use something else to add character: you might insert an instance of something like DaTube into your signal chain in front of Compressor, for example.

All Choice Is Good?

For me, the key lesson from this little experiment is that the differences between the three compressed versions of my example vocal are relatively small when compared with the differences between any one of them and the uncompressed version. Some of those differences between the compressed versions are where that secondary aim (discussed earlier) comes into play, whether it's adding character or wanting more detailed control. If those things are important to you, Cubase clearly has stock plug-ins that let you explore that.

But if you're simply focused on that primary aim of getting your vocal broadly in the right dynamic range, then whichever compressor you go with, be it Vox Comp, Black Valve or Compressor, it can deliver the goods and the 'right' choice will probably come down to how quickly, or how easily, you can get to that good result. Yes, you can deep dive on your vocal compression if you wish, but you can still achieve impressive results by turning a single knob. How cool is that?

We show you how to build a simple sampled instrument in Presence XT.



The first step to sampling your synth is to program a basic run of notes at different velocity levels.

ROBIN VINCENT

Did you know you can build your own sampled instruments in Studio One? Presence XT has an Editor with the power to let you build multisampled, multi-velocity, multi-articulation, scriptable instruments. You may not have encountered it before because it's not included by default; you must buy it as an add-on. However, as many people now receive their PreSonus software products through the Studio One+ subscription, you might find that you already own it.

You'll find Presence XT Editor under Studio One Add-Ons in your online My PreSonus products list.

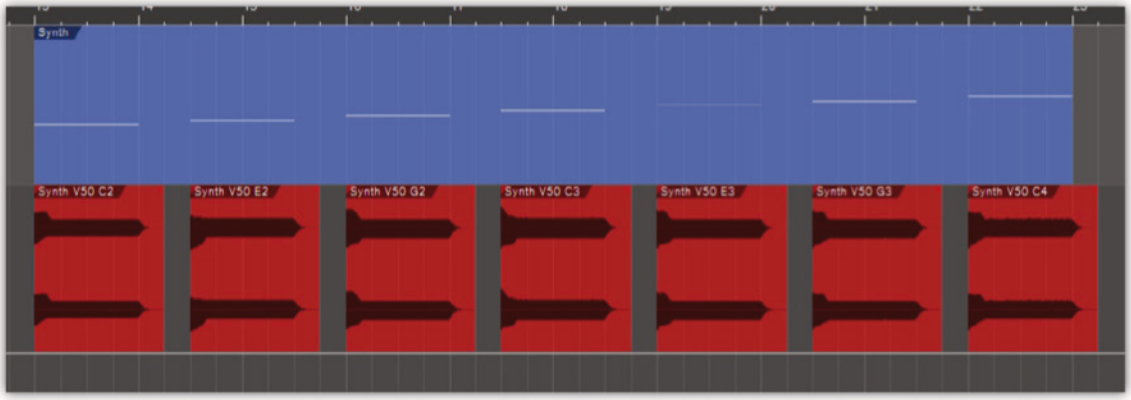
Sampling and building your own software instruments and sound libraries can be an interesting, if sometimes tedious, project. If you've used any orchestral library or other intricately sampled software instrument, then you'll have some idea of the complexity and level of detail these things can run to. There are often multiple samples for every note, in every articulation, with some sustaining, some retriggering and some only happening on the release of a key. There's a whole artform to it, and a world of possibility.

For this workshop we're going to start with the small and simple stuff. Our aim is to capture the sound of our favourite MIDI synth patch at a handful of velocity levels across a couple of octaves. You could ask why we'd do such a thing when we could use the synth as a sound source? There are several reasons why you might sample your synths. You may not have the room to have them set up all the time; this lets you use your favourite sounds without digging the hardware out. It might be a monosynth, but you'd like to play it polyphonically. You might have borrowed it from a friend. Or, for our purposes, it gives us a chance to learn how to make instruments in Presence before we go to the trouble of miking up acoustic instruments.

Taking Notes

To keep this simple, we're not going to sample every note at 127 velocity levels. A modern sampler is quite capable of pitch-shifting to cover a number of notes without it sounding 'sampled'. So, I suggest we sample C, E and G across two octaves. We can sample each note at four velocity levels, to give us some dynamic variation in the sound. The beauty of sampling a MIDI synth is that we can set up the notes and velocity levels in a sequence that will play the synth exactly how we need for the sampling. With an acoustic instrument, you'd have to play the notes yourself with different strengths that you'll have to decide upon and try to replicate consistently with each note. That's not easy to do.

So, in Studio One, create a track for your MIDI synth, create a clip and open the piano-roll editor.



With your note runs recorded, chop the audio file up into individual notes and give your samples suitable names.

When sampling any instrument, you want to give yourself the best chance of capturing something that will be easy to isolate, replicate, edit, sustain and play. You'll have to decide how long you will hold the note. That might depend on how the sound evolves and modulates and whether the held note has a suitable portion that could be looped to sustain the sound. You should also be aware of any release and fade to the sound. I'd recommend turning off any effects on the synth, as these can be added back in later with the effects engine in Presence XT, so you're not waiting on reverb tails or delay taps to fade away into the distance.

With those things in mind, enter the notes C2, E2, G2, C3, E3, G3 and C4 in the piano roll. Leave space between each one to make room for the release and to make slicing up the samples easier. In my example each note is a bar long, and I've put half a bar between each one. Select all the notes and set the velocity down to 32 (25 percent); this will be our first velocity level. Copy and paste the clip, select the notes in the new clip and set the velocity to 64 (50 percent). Do the same for a velocity of 96 (75 percent) and 127 (100 percent). You should now have four runs of notes at four different velocity levels. Great! Let the sampling begin!

Sampling The Goods

Create a new audio track. Plug the output of your synth into your audio interface ready for recording. Hit Record, and the sequence will play the notes and you'll sample the sound of those notes, played by your synth, onto the audio track.

Take the knife (Split) tool and isolate each note. We can always edit them more precisely once we're in the Presence XT Editor. I find it helpful to delete any empty clips around the notes so that everything is nicely defined.

Here's the tedious bit where we have to name all the individual samples. Although I can't find this documented anywhere, it appears that if you put the name of the note into the name of the sample, Presence XT will place it automatically onto the right note in the editor. Coming up with naming conventions is generally a really helpful habit to develop and stick to. So, for our samples, I'm going to give them the patch name, the velocity level and the note name. Right-click each in turn and call it something like 'Synth V32 C2', and then the next note 'Synth V32 E2', and so on. Notes in the next velocity layer would then be labelled 'Synth V64 C2', and so on.

Importing Into Presence XT

Drag Presence XT from the browser to create an instrument track. The Editor exists as a separate page in Presence XT and is accessible from a button at the top right of the front panel.

The Editor has a few different layout view options, but the general gist of it is that at the top you have a Description panel where you can fill in all sorts of details about the instrument and who made it. In the middle, we have our layers and zones, which is where we'll spend most of our time today. Beneath is a bunch of synth-style parameters we can apply to our samples, and with the click of the Wave button, we can edit the sample itself.

To start building our instrument, we need to pull in our samples. We can do that remarkably simply by selecting them all in the arrangement and dragging them into the Zones window. If you've added the note names as I suggested, then with a bit of luck, Presence XT will put all the samples onto the right notes for you. If not, then it will simply start at C1 »

Drag your samples into Presence XT, and if you've included note numbers such as C2, D2 and so on in the sample names, Presence will automatically place your samples on the right keys.

» and spread them about with no regard for the pitch of the actual sample. This is why we name them properly.

Whatever you do, don't at any point click Undo if you make a mistake in the editor: it will unload the instrument and all your editing could be lost. If this does happen, quickly hit Redo to rescue things.

In The Zone

Click on the Grid button to get a visual representation of your sample placement. You should see a blue column for each note. We're not currently seeing the individual samples for each velocity layer; at the moment, they are overlayed on top of each other. To form the basic structure of our instrument, we need to widen the note range of each sample and separate out the velocity layers.

Select the first column and drag the right-hand edge so it cosies up to the next column, effectively extending over notes C#, D and D#. Do this for each column so that each sample covers a number of notes and we have something to play over the whole two octaves. Give it a play with your MIDI keyboard, and you'll hear all four velocity-ranged samples playing together across the notes. So, let's split them up. This is probably done much more easily in the List view, but indulge me because the Grid view is far more visual.

If you click on a column and then click again you should see that the writing is no longer jumbled by the names of all four samples, and you've selected a single sample. In my case, it's the V32 bottom velocity layer. Drag down from the top of the column until the top of the blue box is at a velocity of 32. In the bottom left corner is a list of parameters, one of which is 'High Vel'. That's where it needs to say '32'. Click twice in the column above the V32 square and you'll select the V64 sample. Drag the top down to 64 and the bottom up to 33. The same applies to the next two samples, setting them to their respective velocity layers. If you lose track of a sample, it's probably stacked with another one. Check the List view and you should be able to locate it. Don't undo!

Now you've got the hang of doing one column, we can save time by doing the rest in one swoop. Click twice in the next column to select that first sample, then hold Shift

With your samples imported, go to Grid view and drag them out so that they cover the correct note and velocity ranges.

and click in the last column to select all the samples in between. It should select the same sample in each column. Then pull the tops and bottoms to set the right velocity for each, and they'll all move together. You should now have a grid of note-mapped and velocity-layered samples that offer a certain amount of dynamics to your playing.

In practice, I find it quite hard to trigger the V32 samples; I can't play gently enough. So, maybe it would be better to use a single sample all the way up to a velocity of 64 and then have more variation in the top half, where it might be more effective. You could set this up with the samples we already have or, to do it more properly, you could go back and sample your synthesizer at a different range of velocities.

Sustain

One last thing to bring our freshly made instrument to a sense of completion. When you strike a note, it only plays for the duration of the sample and then stops. What would be nice would be to have the note sustained indefinitely while we are holding the key. We can do this by setting loop points on each sample.

Select a sample in the grid and click the Wave button. At the bottom, you'll see your sample laid out, with a few parameters on the left. Above the waveform window is the name of the sample. Alongside the name is a little play button, which will play the sample as if you've played it with the keyboard. In the parameters section, turn the Loop Mode to Sustain. A loop range will appear above the sample, and what we're trying to do is find a section of the sample that loops in such a way that you can't hear the join. So, drag in the handles in from the left and right and see what you can find. The success of this will greatly depend on your sample and the sort of effect you're after. You can try to do all your samples at once, but I've found that you can't rely on it to be the same point for every sample.

Don't forget to save your preset by clicking the tiny little button that looks like a sheet of lined paper at the top left of the interface, next to the bypass button. There doesn't appear to be any way to create a user folder or anything, so you'll just have to go with the categories that are there already. If anyone finds out a way to make your own, let me know!

Presence XT lets you set loop points for each sample, so that they can sustain indefinitely while you hold the note down.

There's plenty more that Presence XT can do to help you craft interesting, sampled instruments. I hope this introduction to the Presence XT Editor has given you a starting point.

DPA MICROPHONES

DDK4000
Drum Microphone Kit

DPA's selection of mics for the drum kit provides the most natural and precise sound possible, giving you complete control.

dpamicrophones.com

Spotlight

A tracking session for Toontrack's Superior Drummer 3

No room for a drum kit? No problem!

LUKE WOOD

Recording acoustic drums can be one of the most challenging tasks for many home and project studio owners. Even with a healthy selection of microphones and plenty of inputs to hand, there's no getting around the fact that drums generally need a good amount of space around them — as well as decent acoustics — to really sound at their best. Thankfully, those of us without spacious, luxurious live rooms are spoilt for choice when it comes to virtual drums, with many manufactures offering the sound of some of the world's best studios and players right inside of our DAWs.

BFD3

The ever-popular BFD is now in its third iteration, and has been designed to cater to those after instant mix-ready drums as well

as users who prefer to dig a bit deeper into crafting their sounds. There are seven kits in the factory library, which includes dedicated rock, metal, jazz and brush variants, with two different recording locations offering some contrasting room sounds, while a comprehensive mixer comes loaded with presets along with modelled processors and multi-output routing. Although the instrument is primarily sample-based, there's also some modelling in play, with parameters such as bleed and tom resonance adding some additional flexibility that goes beyond what sampling alone can achieve. A collection of grooves are included, and a wealth of additional patterns, kits and extra individual drums can be added thanks to a collection of expansion packs. If you're only after the basics, it'd be well worth a look at the company's BFD Player, which combines a 5GB library with 340 drum patterns and a simplified mixer — best of all, it's free!

$ $329.99.

W www.soundonsound.com/reviews/fxpansion-bfd3

W www.bfddrums.com

DW Soundworks

Drum Workshop (DW) are well known for their acoustic kits, but earlier in 2024 they took their first leap into software with DW Soundworks. Although it was announced

alongside a new line of convertible acoustic/electric drum kits, it can be used separately in the same way as any conventional drum library. As you might expect, there's a healthy dose of DW kits present (from the Collector's Series) but you also get a Gretsch USA Custom along with DW founder Don Lombardi's personal 1958 Slingerland kit, and the company say that more will be added through future updates. In terms of processing, EQ, compression and reverb can be applied to individual tracks or across the whole kit, and there's a built-in library loaded with grooves that cover a variety of musical styles.

$ $199.

W www.dwdrums.com/products/deswsoundworks-dw-soundworks

EastWest Pro Drummer

Spread across two instalments, Volume 1 and Volume 2, EastWest's ProDrummer contains over 100GB of high-quality drums played in five different studios by Steven Sidelnyk and Matt Chamberlain, with mixing and production duties taken care of by Mark 'Spike' Stent, Joe Chiccarelli and EastWest's own Doug Rogers. Plenty of control and processing is on offer thanks to a 16-channel mixer, which comes loaded with authentic emulations of SSL's famous EQ, transient designer and compressor modules and a built-in reverb — there's even an amp simulator for carrying out more heavy-handed processing. For those who want to keep things simple, the collection comes packed with user-adjustable presets that offer a range of track-ready songs right out of the box. A staggering 14,000 MIDI grooves are included, and a Songbuilder window allows users to arrange their entire song in ProDrummer, as well as allowing parts to be dragged and dropped directly onto a DAW track.

Virtual Drums

$ Each volume $199, Bundle $299.

W www.soundonsound.com/reviews/eastwest-prodrummer

W www.soundsonline.com/drums-and-percussion/prodrummer

GoranGrooves Handy Drums

Handy Drums is designed to offer users a simple and intuitive way to add great-sounding drums to their tracks, with a stripped-back GUI providing quick access to the essential controls. Despite the apparent simplicity, there's plenty going on behind the scenes, with carefully recorded and processed samples promising to deliver the most realistic sound possible — GoranGrooves say their variable hi-hat feature offers the best sound you'll find in any virtual drum instrument! As of the latest version, tuning is now available on a per-instrument basis, and a dedicated mode has been added to improve the software's response when being triggered from an electronic kit. A browser offers access not only to the company's own MIDI packs, but also any third-party MIDI files on the user's computer, and a mapping system makes quick work of creating and saving custom MIDI mappings. A wide variety of kits is available and can be purchased either individually or as part of a larger bundle.

$ From $20 each, Producer Collection $249.
W www.soundonsound.com/reviews/goran-grooves-handy-drums
W library.gorangrooves.com/virtual-instrument-plugins

Hertz Drums

The latest version of Hertz Drums comes loaded with a 10-piece kit that boasts up to eight independent articulations for every drum, with separate recordings of left- and right-handed hits. A MIDI groove player makes light work of developing ideas, and over 1560 patterns and fills played by real drummers are included as standard, all categorised by song section. As well as offering a wealth of additional kits via expansion packs, the software includes a fully functional sampler and extra drum pads for incorporating custom samples, and there's also support for a number of pre-configured maps based on other popular virtual drum products, so that Hertz Drums can be integrated seamlessly into existing sessions.

$ $49, expansions from $89.
W www.soundonsound.com/reviews/hertz-drums-bundle
W hertzinstruments.com

Native Instruments Studio Drummer

Recorded at Teldex Studios in Berlin, Studio Drummer comes loaded with Pearl, Yamaha and Sonor kits, along with cymbals from the likes of Zildjian, Sabian and Paiste; additional Mapex, Ludwig and Masshoff snares are then joined by a handy selection of percussion including tambourines, sticks and cowbells. The library's mixer offers a variety of close and distant mic signals, and boasts built-in tape saturation and convolution reverb, as well as NI's renowned Solid Bus Comp, G-EQ and Transient Master processors. Over 3300 grooves played by top pro drummers are included, with variable Tightness and Swing controls allowing users to adjust the overall 'feel' of their drum parts. As well as providing mapping compatibility with a range of other popular drum libraries, Studio Drummer is also loaded with templates for Roland's V-Drums range of electronic kits.

$ $149.
W www.native-instruments.com/en/products/komplete/drums/studio-drummer

Native Instruments Abbey Road

Available either individually or as a full bundle, NI's Abbey Road range spans six eras of drum sounds, all captured at London's legendary Abbey Road Studios. As you might expect, the collection shares much of its functionality with Studio Drummer, but with each instalment recorded using period-authentic equipment and techniques. Vintage Drummer offers the sound of the 1930s and 1940s, while 50s Drummer covers the early days of rock & roll along with jazz, country and rhythm & blues. 60s and 70s Drummer continue to follow the evolution of recording gear and practices before 80s Drummer tackles the era of power ballads and gated reverb. Finally, Modern Drummer brings things back to the present day, employing a mix of the legendary studio's modern and vintage gear. To further assist users in crafting authentic-sounding parts, each library also includes a collection of MIDI grooves played in the style of its era.

$ Individual $99, Bundle $199.
W www.soundonsound.com/reviews/native-instruments-abbey-road-vintage-drummer
W www.soundonsound.com/reviews/native-instruments-abbey-road-50s-drummer
W www.native-instruments.com/en/products/komplete/drums/abbey-road-drummer-collection

Spitfire Audio The Grange

Spitfire may be best known for their large orchestral libraries and experimental offerings, but their product range also covers the likes of pianos, guitars and — you guessed it — drums. The Grange brings together three high-profile rock drummers (Roger Taylor, Chad Smith and Andy Gangadeen) and captures a huge collection of hits and loops at Headley Grange, a studio that has played host to acts including Led Zeppelin, Fleetwood Mac, Genesis and many more. Close mics for kick, snare, hi-hats and toms are joined by four stereo signals captured at increasing distances, covering everything from tight, dry-sounding kits to more spacious, room-heavy sounds, and those who want instant results can choose from a selection of three pre-mixed Tight, Mid and Epic signals (or a fourth Super Tight option, in the case of Andy's kit). A built-in loop browser makes quick work of finding the right rhythm for the task at hand, and the library also features a Kickstart engine designed to quickly map patches to any MIDI controller.

$ $299.
W www.soundonsound.com/reviews/spitfire-audio-grange
W www.spitfireaudio.com/virtual-instruments/the-grange

» Steven Slate Drums 5.5

Steven Slate Drums promises to deliver the most versatile range of drum sounds available in a virtual instrument, with an array of kits and artist presets covering jazz, funk, reggae, rock, metal and much more. Along with its wealth of sounds the library includes over 2400 MIDI grooves housed in a dedicated browser, and if that's not enough, there's the option of importing third-party files. Integration with MIDI controllers and electric kits is also made simple thanks to some intuitive MIDI Learn and Map features. The software's mixer section provides in-depth control over the levels, dynamics, panning and ADSR setting for each individual kit element, with close mics joined by a selection of room and ambient signals. If you're happy to stick with the basics, the company also offer a free version that includes a single kit and a collection of pre-mixed presets.

$ $119.99.

W stevenslatedrums.com

Toontrack EZdrummer 3

Designed specifically for songwriters, EZdrummer 3 combines a collection of high-quality drum samples with a wealth of built-in writing, editing and mixing tools. There are seven kits that can be customised with additional kick, snare and cymbal options — you get a healthy dose of percussion thrown in too — and three different recording rooms offer varying levels of ambience. A grid editor and timeline make it possible to create, edit and arrange parts without heading back to your DAW tracks, and there's the option of balancing and processing either internally or externally thanks to a mixer with 16 stereo outputs. You get a MIDI library packed with over 2500 grooves and fills played by real drummers, all of which are categorised by musical style and song section, and a Tap2Find feature makes it possible to 'play in' ideas and have EZdrummer 3 suggest a list of matching parts. A Bandmate function then takes things a step further, analysing MIDI or audio files of existing instrument recordings and generating a customisable, matching drum part. For those who want more, an ever-expanding collection of EZX add-on packs offer a huge range of additional kits and MIDI parts that span a vast range of musical genres.

$ $179. Expansions $89.

W www.soundonsound.com/reviews/toontrack-ezdrummer-3

W https://www.toontrack.com/product/ezdrummer-3

Toontrack Superior Drummer 3

Toontrack's flagship drum instrument steps things up a gear from EZdrummer, with a 230GB factory library captured by legendary engineer George Massenburg. To offer some tonal variation, a selection of the seven kits have been played with brushes, rods and felt mallets as well as sticks, and there's also a collection of electronic drum sounds to complement the acoustic kit samples. The software boasts a comprehensive built-in mixer kitted out with 35 insert effects and a huge range of presets created by Toontrack's in-house team and world-class mix engineers; precise control of the overall sound is available courtesy of a wealth of individual mic signals and bleed controls, and output formats from stereo up to 11.1 surround are supported. Along with its editing and arranging facilities, Superior Drummer 3 also includes the Tracker, an innovative feature aimed at augmenting or replacing existing drums that can create new parts based on existing MIDI files or audio recordings. Superior Drummer can be expanded further thanks to a mind-boggling array of SDX packs, which offer kits, rooms and MIDI content covering just about everything from 1920s-style vintage kits to hard-hitting modern metal setups.

$ $399, expansions $179.

W www.soundonsound.com/reviews/toontrack-superior-drummer-3

W www.toontrack.com/product/superior-drummer-3

UJAM Virtual Drummer

UJAM's series of Virtual Drummer instruments offers a wide variety of drum sounds and musical styles that can be purchased individually, or as part of a larger bundle. The company pride themselves on creating 'frustration-free' software that promises to yield fast, high-quality results for beginners and pros alike, and the range covers everything from vintage '60s and '70s sounds to hard-hitting modern rock and metal, as well as just about everything in between. Each instalment includes five extensively sampled kits, a set of mix-ready presets and a variety of drum patterns that can be fine-tuned with a range of performance controls. As well as offering a great sound right out of the box, Virtual Drummer also provides plenty of in-depth control over each kit's balance and tuning options, and supports multiple outputs for applying further processing in a DAW.

$ Individual $99, Drummer's Bundle $299.

W www.soundonsound.com/reviews/ujam-virtual-drummer-brute

W www.soundonsound.com/reviews/ujam-virtual-drummer-deep

W www.ujam.com/drummer

XLN Audio Addictive Drums 2

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How I Got THAT SOUND

Andy Jackson ▪ Pink Floyd 'Marooned Jam'

JOE MATERA

English recording engineer Andy Jackson has worked with Pink Floyd and David Gilmour since 1980, having previously mixed the Boomtown Rats' number one hit 'I Don't Like Mondays'. He is also an artist in his own right, and his latest solo album *AI AJ* was released in February 2024; check out Andy's website at https://andyjacksonmusic.com for more information. A favourite among the many great sounds Jackson has recorded over the years is the dry, dense drum sound heard on Pink Floyd's 'Marooned Jam' from *The Later Years 1987-2019*.

Rocking The Boat

"We started off in David Gilmour's Astoria houseboat studio, assembling ideas and knocking them about, with the band set up in the studio there, which was this tiny room, about four metres square. It was so tight you couldn't move in there. Due to space, Nick [*Mason*] set up a small kit, and I miked it with just the basics, four mics on the drums: an AKG D12 on the kick, a [*Shure*] SM57 on the snare and a couple of [*Neumann*] U87s as overheads. I didn't mic the toms because two more mic stands were more than there was room for. I added a couple of mics on the Leslie [*a Neumann KM84 on the top and a Sennheiser MD421 on the bottom*], DIs for the keyboards and bass and a couple of U87s on David's guitar. We recorded a whole bunch of stuff as demos, working out arrangements and things like that until we got to the point where we knew what the record was going to be, so we started recording it.

"We moved to Olympic Studios and cut a drum track for 'Marooned Jam', but we all agreed we preferred the drum sound recorded at the houseboat. We all thought it had a sort of solidity and dryness to it which was just much more suitable to the music. So, we went back to Astoria and we redid the recording, apart from a couple of bits of keyboards and the orchestra, in that tiny room. Nick set up a bit of a bigger kit, but essentially, it's the same thing. He was just rammed into a corner in this tiny room, he could scarcely get in and out of the kit. There was just enough room to get out, and one of the mics was backed entirely into the wall along with one of the overheads. It was just incredibly squashed in, but it really gave it that sound, which is incredibly dense and really dry.

"Any reverberation you hear was artificial as it was added on later, basically just an Echoplate. I did cheat on that mix and snuck in some close tom-tom mic samples to get some bottom end, because they weren't miked, it was only on the overheads. I love the sound because it's got a sort of neutrality to it. There's no room tone on it at all. The room's very dead. And we killed it further because of the fact they were all in there. We added some foam above the drums and things like that to try and get rid of any ceiling reflections. And it's almost as if you were recording the drums in a vacuum kind of thing.

"This same drum setup used on 'Marooned Jam' was also used on 'Louder Than Words', which appeared on Pink Floyd's 2015 album *The Endless River*." ■■■

Hear The Sound

W www.youtube.com/watch?v=AWt6Hiyvtyo
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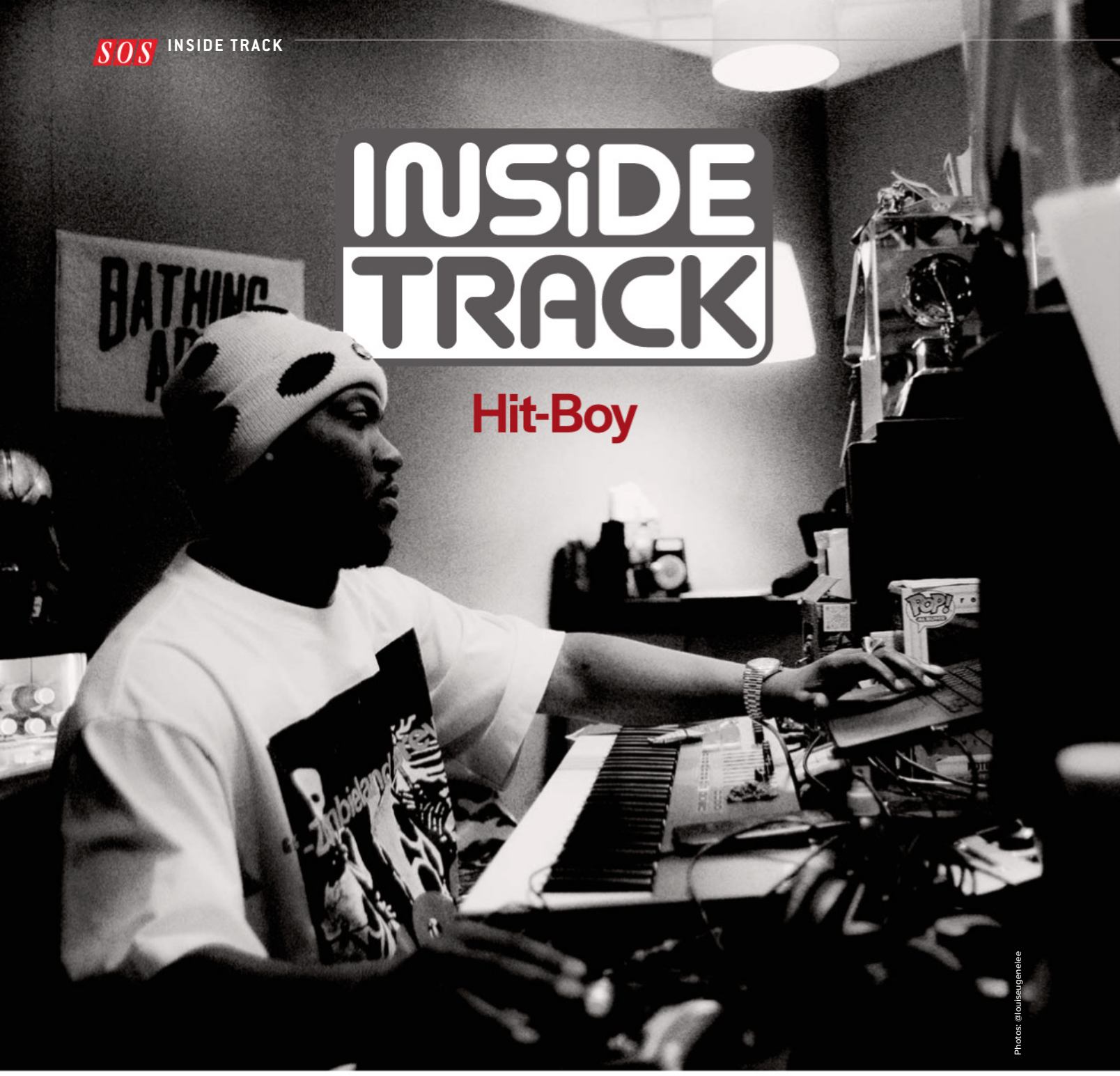
INSIDE TRACK

Hit-Boy



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PAUL TINGEN

Hit-Boy's freewheeling production style has made him one of the hottest producers in the USA.

"My process is not having a process," insists Chauncey Hollis Jr, aka Hit-Boy. "I don't get too industry about making beats. I make my best stuff when I'm just free, when I'm going off no-one's notes, going off no references, when nobody is putting me in a box or pigeonholing me in what they think the vision is. That's why I say I'm the least technical producer ever. I just do it, you know what I mean? And the stuff that I feel in my soul, people pretty much like it.

"It's crazy, because you don't know what things turn out to be. You might not feel the most connected to a big hit you've been involved in. I've made songs that are all over the radio. Not one of the songs I did with Nas made it to the radio. But I've gotten more respect, been more revered, with more people in the streets telling me how much the albums I did with Nas inspired them, than with many of my big hits."

Clearly, there's more to Hit-Boy's music career than having hits. Since 2020, he's released six albums with Nas: *King's Disease I*, *II* and *III* and *Magic 1*, *2* and *3*, plus many more with other artists or under his own name. The *King's Disease* albums have received two Grammy

nominations and a Grammy Award, as well as contributing to Hit-Boy's two Grammy nominations for Producer Of The Year. His most recent nomination was on the basis of releases by Don Toliver, Dreamville, Musiq Soulchild, Nas, his solo albums *Surf Or Drown Vol 1* and *2*, and releases with the Alchemist.

Hit-Boy's hectic release schedule continues this year with an album with regular collaborator the Alchemist and his father, Big Hit, called *Black And Whites* (see box). Meanwhile, his writing and production work with singles for others tend to yield the biggest hits, including Kanye and Jay-Z's seminal 'N****s in Paris' (2011), Travis Scott's 'Sicko Mode' (2018), and songs for Drake, Kendrick Lamar and Beyonce. This year, he's already enjoyed a synth credit on Beyonce's megahit 'Texas Hold 'Em', and co-wrote and co-produced Jennifer Lopez's hit single 'Can't Get Enough'.

Brave Beats

Hit-Boy grew up in Fontana, part of the Inland Empire and an hour's drive outside LA, which is why he also calls himself Tony Fontana. He had his first breakthrough in 2007, on Jennifer Lopez's hit album *Brave*. "I sold a beat to Jennifer Lopez for a song called 'Forever' that appeared on *Brave*. I had turned 20 years old and starting on a pop level like that opened up my eyes. It was like, 'Man, I can do anything!'"

He had started making beats only a few years earlier. "In 2001, I was rapping, and after a year also started making beats. A friend of mine had a setup with FL Studio, or Fruity Loops as it was called. I thought it was fun and started messing with it. He also had this program called [*Sonic Foundry*] Acid Pro, where you could chop samples and record vocals. It was like Pro Tools back in the day. He had all the tools, and I kind of peeked at them and ran with it. I had all the pieces I needed to make albums, songs and beats from the jump. I did not have an MPC, because I came in when it was time for a shift. But people did not really take FL Studio serious when it first came out. It was like, 'Oh, that's some kid's stuff,' or 'It's like a game.' It took time before people made big hits with FL.

"Thank God for my family, because when I decided that I wanted my own computer, to be able to make beats every day, everybody was pitching in on my vision. My grandmother let me have a desktop computer with a screen that she already had, my uncle bought me a basic microphone and an [*Digidesign*] Mbox, and my mom bought me some speakers. She was happy that I was in the house making music instead of being in the streets!"

Doing It All

"For many years I had a studio in my bedroom and invited people to my house to make music with me. Hundreds of artists came through to rock with me before I did anything professional. I used to have everybody who came through sign with a marker on my wall, and ended up with a wall full of signatures. I was inspired by everybody, from Jay-Z to Kanye West, people on the West Coast, listening to gangsta rap, Snoop Dogg, Ice Cube, NWA, all of that stuff.

Hit-Boy: "I got an email from Kanye saying, 'Yo, me and Jay just made a song to one of your beats. I promise your life is going to change after this song drops.'"

"I always loved music in general, which is why people don't really know me for a specific sound. I make everything under the sun, whether it's a West Coast vibe, or East Coast. I'm able to do six albums for Nas, but then worked with a new group from the West Coast called BlueBucksClan, you know what I mean? I've allowed my palette to build and be exactly what I want it to be, which is anything. I pretty much do it all.

"A lot of people like my drum programming, melodies and stuff like that. But I'm primarily a programmer. I learned many chords and in 2006 got to be around a lot of gospel musicians and dope musicians, like 1500 Or Nothin'. It was about being in the right circles and studios, and learning from them, and from everybody who was a super musician. I never made it to the point where I can play an instrument on stage, but I play well enough to make beats, and understand melody.

"I still make my beats in FL today. I've been using it for so long, it's an extension of me at this point. We just flow together. I had times back in maybe 2005 where I tried Logic, and I tried Ableton back in 2015-16. But my pace and my workflow is just better on FL. It's so visual, and I'm so used to it. I also remained in my home studio for many years, until I moved to a professional recording studio in West Hollywood called Chalice about six years ago, and refocused my energy. It's a real, smaller space, also with a homey vibe."

French Fancy

His ability to be "anything" has clearly served Hit-Boy in becoming one of the leading hip-hop/R&B producers of his generation, with a credit list that also includes Big Sean, A$AP Rocky, Kid Cudi, John Legend, Mariah Carey, Nicki Minaj, Selena Gomez, Rihanna, Ariana Grande, Juice Wrld, Jack Harlow, 2 Chainz, the Game and many others. After getting the placement on *Brave*, Hit-Boy became a staff producer for Polow Da Don's Zone 4 label, receiving co-production credits on the *More Than A Game* documentary soundtrack, and on the Mary J Blige song 'Stronger'. In 2011, he signed to Kanye West's GOOD Music label, and immediately hit the super-bigtime because of his involvement in Kanye and Jay-Z's megahit 'N****s In Paris', which won two Grammy Awards.

"The 'Stronger' song was a good moment. To be working with Kanye was unbelievable. Just to be around somebody like him, who I had been idolising, watching, respecting, listening to, being inspired by. Just for him to want me around. We spent a lot of time together, and we worked on many songs and ideas. The 'Paris' beat was probably up when we first started working, but he was trying to give it to Pusha T, who just couldn't think of anything to it. Pusha said it sounded like a video game to him.

"This was funny because that's how I look at my beat-making process. It's always been fun for me. I never really go too heavy on the technical side, even to this day. It's more all instinct and what I'm feeling in the moment. I might lay a chord progression that I might not remember how to play afterwards. With 'Paris' I had come up with the musical bed. Then Mike Dean and Kanye layered over what I had done. »

Although he's now moved to a dedicated space at Chalice Studios, Hit-Boy still works mainly with a simple rig based around Image Line's FL Studio.

» "Later I got an email from Kanye saying, 'Yo, me and Jay just made a song to one of your beats. I promise your life is going to change after this song drops.' He kept hyping it up through email, telling me how great it was. The first time I actually heard it was at the listening party for *Watch The Throne* in 2011. It was a real experience. When the first song played, people clapped, when the second song played, everybody went crazy. But when 'Paris' came on, the whole building stood up. It went nuclear! So I knew the song was a different level."

Beat Development

'Paris' took Hit-Boy's career to a different level, and he followed up with several more classic hits, among them 'Clique' (2012) by West, Jay-Z and Big Sean, 'Backstreet Freestyle' (2013) by Kendrick Lamar and 'Sicko Mode' (2018) by Travis Scott. "'Clique', 'Paris', 'Goldie' and 'One Train' by A$AP Rocky, 'Backseat Freestyle', all started from beats I made at my mom's house in the Inland Empire when I was by myself, just working out of my bedroom. That's why I've been super productive, selling hundreds of beats, doing six albums for Nas, doing albums for Musiq Soulchild, albums for BlueBucksClan, Benny the Butcher, and so on.

"It's funny because I've been saving all my FL files since '06-'07. Sometimes I'll go back and I literally will remember exactly where I was when making certain beats and how I felt at that time. All these memories come back to me. I number all my beats, so it'll be like 'Hit-Boy 1873'. I take these old beats, and sometimes I still sell them, or I update them with the new skill sets that I've developed, just to put the new knowledge into it. Or, if I used a sample that doesn't get cleared, I replicate the feeling and the sound without samples.

"The original beat for 'Clique' was actually called 'CDC' and released by Dom Kennedy [*in 2011*]. We changed it up, messed with the melody, brought that bass line in, and took it to a new level. With Kendrick, I worked with him on *Good Kid, M.A.A.D City* [*2012*], and together with Dave Free we went through some beats of mine, and when he heard this one, he was like 'Yeah, that's it.' I still see clips of Kendrick performing that at his concerts.

"'Sicko Mode' started with something that Rogét Chahayed and I produced. I played it to Travis, who loved it. I had actually worked on his album before, *Birds In The Trap Sing McKnight* [*2016*], and the night that album dropped, Travis told me, 'Yo, I'm in the studio with Drake. Can you send me that "Sicko Mode" beat?' Drake hopped on it immediately, and laid down his part for the intro. Next thing I knew, Travis had pieced it together with two other songs and made it a whole moment."

Streamlining The Studio

Although he's upgraded his surroundings, Hit-Boy still works on gear that's as basic as it was two decades ago. "I still use FL Studio on a PC. I went through parts of my career where I had FL on the Mac, but I just feel like the sound, the way it feels, it just hits harder on a PC. I have a Universal Audio Apollo interface, and an FL Studio MIDI keyboard. Sometimes I'll use an [*Akai*] MPC Live Supreme, but more to record live audio back into FL Studio, and I then manipulate it in there.

"My monitors are Focals and QSC live monitors. I couldn't tell you the exact Focal model, but they're mid-tier ones. They're for fine-tuning and the QSCs are for vibing, when I want it loud and I want to make sure that my 808s aren't going to tear through the speakers and

break them. You can beat the QSCs up and be loud as you want without the speaker busting. When I want to listen at low volume and really get the details, I put it on the Focals.

"With plug-ins and VSTs I also keep it simple. I have Spectrasonics Omnisphere and Trillian, and the Native Instruments Kontakt Player, and stuff like that. Kontakt has new modern ways of thinking and modern sounds. I also use a plug-in by StudioLinked called OVO RnB, and the Juice plug-in by Gospel Producers. I'm developing my own plug-in with StudioLinked. It should allow you to make something dope without even trying. I feel like that's where it's going. I want to be able to have a VST where anybody can just press some keys and make it sound great.

"For things like EQs, I use stock FL Studio plug-ins. I also use the Valhalla Vintage Verb, and a few other things. I also have the Serato sampler. I'm always looking for new sounds and have been collecting samples since 2003. I have millions of samples, and will shuffle through stuff that sounds interesting and see how I can make it my own. I'm organised and unorganised at the same time. I just find samples in the moment. That's what keeps my ideas fresh. I look out for old sounds that I might have missed that I already have, or I might look up a certain type of kit online and apply that to what I'm doing in the moment. I'll do everything from speeding the sample up, slowing it down, pitching it. »

Black & Whites

Hit-Boy began work on *Black & Whites* immediately after Big Hit came out of prison, in May 2023. "The moment my dad was able to get out of prison and come back to California, we messed with 'Only Weight I Feel'. I wanted to set a tone for the album before we went to talk to Alchemist. I wanted to have records that Alchemist could hear and be inspired by. I thought that beat was special and going to set the aesthetic for the album.

"Anything that's grimy street and sounds authentic is going to work for my dad. So it's a stripped-down beat, just sample, drums and bass line. The sample came from a movie score or a TV show score. I stripped it down and reimagined it in FL. I think I used Serato to take some sounds out of the sample, and just built around that. I always try to find a pattern and some movement just with the sample. After that I layered some simple drums on top that are still unique. The bass in this track was a one-shot acoustic bass, because it's very deep. There's no 808."

After creating 'Only Weight I Feel', Hit-Boy, the Alchemist and Big Hit worked together as a trio, with the former two creating the beats and the latter adding his raps. "That was pretty much the vibe. We went to Alchemist's studio and locked in. Alchemist also came to our studio. It was maybe a month's process of recording and getting the beats and the interludes and segues together.

"I think Alchemist uses an MPC 2000 XL, and pulled up ideas that he or I already had, and imported from FL or the MPC into Pro Tools. Alchemist and I have a lot of output between the two of us. We used a lot stuff that we had stacked up. And we added a few beats. We'd then work on the structure of the songs, just to put them in a format that's digestible. Most of the stuff I didn't record myself, but I did vocal production and just moving stuff around, flying hooks to different places, and vocal producing."

» "With 808s, I switch it up. If you listen to the song 'Another One' [*on the 2024 benefit album* Rent Due *by La Russell and Hit-Boy*], or 'ALLAT' [*2024, Hit-Boy, Big Hit & Zoe Osama featuring Honcho*] you'll hear this kind of modern sound. It's almost not an 808, and more like a Moog sound. That's a new sound I've been running with lately. When I was coming up, my favourite producers, from the Neptunes to Lil Jon to Timbaland, would use signature sounds. The Moog-like 808 sound is deep and hits like an 808, but it has this top that gives it a different bounce and a different feel."

Growing

While Hit-Boy likes to hark back to his beginning days as a beatmaker, he's also developed other skills, not only as a producer, but also as an engineer, notably on the albums he has done with Nas.

"Nas and I dropped like 80 songs in three years. I was pulling from every piece of energy and every beat I had. Some days I start by figuring out chords or figuring out a melody, or I might hear a sample. Or it might be drums. A lot of my ideas or just my inspiration come from pulling up old samples, old drum patterns and laying new samples over my old stuff. I had to reproduce a lot of stuff, too, because with Nas it's a soulful hip-hop vibe.

"That has allowed me to grow as a producer, and as a music maker. When I work with Nas, I'm definitely going to have a pack of beats ready to play him. Nine and a half times out of 10, he's going to try an idea to something I played him, but sometimes he sends me samples or tells me a certain vibe he wants to talk about, and I'll kind of get into that or bring a piece of music that will complement that. And I might help him arrange a song. We always have open lines of communication.

In session with BlueBucksClan at Chalice.

"Nas came to my studio at Chalice. Me being able to record and vocal produce him means it's a quicker process, versus me having to wait for the engineer to do this. I used a Sony C800 mic, and [*Neve*] 1073 compressor and the Tube-Tech CL-1B [*compressor*]. It's a simple setup. That C800 cuts clean through. You're gonna feel the vocal and you're going to hear it clearly. I record him into Pro Tools, and then do the vocal production, 100 percent. That's why I feel like I'm a better beat maker and a better producer, because when I work with people like Nas or Benny [*the Butcher*] I get into the vocal production and find places for hooks and try to fly different parts around.

"Once I get a song in a good place with a rough mix, I'll sit with my main mixer, David Kim, and finalise it. We clean the song up in my Pro Tools session, and get it to a ballpark idea of where I need it to be, and then he'll take it to his studio in the back of Chalice and do his final mix. He's going to make sure that everything is blasting out at the right levels and hitting the way I want it to, with the clarity at the same time. It's a simple process."

Alchemy

Recent projects released under Hit-Boy's own name include two albums with Big Hit, *The Truth Is In My Eyes* and *Paisley Dreams* (also with the Game), both self-financed and released on his own Surf Club label. Hit-Boy also split production duties with star producer the Alchemist on Benny the Butcher's album *Everybody Can't Go*. By the time this interview is published, *Black & Whites*, the collaborative album by Hit-Boy with Big Hit and the Alchemist, will have seen the light of day (see box).

"I flew a few times to Las Vegas to work with Benny. I did some records with him, and then Alchemist came in and brought the full vision to life. The Alchemist and I really respect each other. We've been through similar stuff with the industry, having to completely empower ourselves and take the independent route and work with artists that truly want to craft full projects, or craft like a special art piece, versus chasing a hit. Most of the stuff I listen to is underground stuff that's not even really mixed or mastered properly. It's like a raw feeling, raw energy. And that allows me to know, 'OK, you don't have to be perfect out here.' Just make something that you really feel and let it fly."

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TalkBack

Leslie Gaston-Bird

WILLIAM STOKES

Leslie Gaston-Bird's CV boasts an unusually broad array of career exploits, from her role as Broadcast Recording Technician with America's national radio station NPR to recording the Colorado Symphony at the famous Boettcher Hall. "I worked at a radio station while I was at uni," she explains. "When I got my job at National Public Radio, they gave me a 31-question exam, which had to do with things like, 'How does a compressor work?' 'What is a preamp?' 'What is the input and output impedance of a console?' And I knew all the answers to those questions. So I got the job!"

Today, Gaston-Bird is the owner of Brighton's Golden Reel Award-nominated audio post-production studio, Mix Messiah Productions, President of the Audio Engineering Society and a lecturer in Sound Recording and Music Production at City University, London, having completed her PhD not long ago. She's also written the Routledge-published book *Women In Audio*, profiling almost 100 notable female practitioners across eight chapters representing different disciplines.

"You've gotta have five jobs at a time!" She laughs. "But now my goal is to just have one job..."

At the moment I can't stop listening to

Well, at the moment I can't stop listening to student projects, because I'm marking! I've had Thundercat on loop, pretty hardcore, particularly while I was doing my PhD. His album *Drunk*. And then, Pat Metheny and Lyle Mays did the album ages ago called *Still Life (Talking)*. So I listened to *Still Life (Talking)* on repeat while I was working on my PhD because it soothed me, it calmed me down. More recently, in the last three days, I've been listening to MC Solaar's *Paradisiaque*. Lately, it's been jazz, hip-hop. I mean, the bass lines... I like that fretless bass sound. As I was listening to *Still Life (Talking)*, I went and looked for a live performance of one of those songs. You know how some people say, "You can't make it sound like the album on stage"? Well, actually, you can. The Pat Metheny group: my God. To see it performed live only added to it. I had that on loop. I was calling my son, like, "Look at this! Look at the marimba player! Look at the guitarist putting down their guitar to play the marimba! Oh, my God! That's amazing!" I found out one of the guitarists passed away, but I would have loved to get to know the guy. He sang, played guitar and marimba. All in the same song!

The project I'm most proud of

Well, I would say that it was [*2023 movie*] *Publish Or Perish*, which I did the sound for. We just got back from Los Angeles, where we were nominated for a Golden Reel award. So I'm very proud of that Golden Reel nomination. Especially given that we had such a small crew, and we lost out to *The Last Kingdom*, which had a huge budget! But you know, we were in that same category. And I'm very proud of that.

David Liban directed the film. It's about a professor who accidentally runs over a student with a car, so we're gonna say that's a dark comedy... And this professor wanted tenure. I don't think you can get tenure in England; but in America, if you are a professor and you get tenure, it's really hard for them to fire you. So the professor in this movie is like, "I gotta get tenure," and he's trying to turn in his dossier and he runs over this kid. And he's like, "I'm gonna take you to the hospital as soon as I drop off my dossier," but the kid dies. So it asks how you live with yourself after that. And it's funny, so please watch it! I can relate to the story, just having been under that kind of pressure. I never killed anybody, though!

I did the re-recording mixing for that. Our little crew put this thing together. And to do it in a small studio like mine, in 5.1 surround sound, using iZotope and all the tools that we have access to, being trusted by a director, doing the final mix... It just sort of validated me, it was like, "Yeah, I can be trusted by a director to do great sound design." We wanted to finish the film, but we could take our time with it. And I really enjoyed the process of doing sound design for that film.

The first thing I look for in a studio

The first thing I do when I walk into the studio is this: [*claps*]. I like it when it's on the dry side. One studio I've been in love with was at National Public Radio in their old building: Studio One, I think, at the Massachusetts Avenue [*Washington, DC*] building. It was clear as a bell. Transparent. Lovely. You know, a studio can look pretty, you can see the cosmetics and you can see the treatment, but the question is: how does it sound? I want to know how it sounds.

The person I would consider my mentor

I'm gonna say Wayne Jackson. And David Pickett. Those were the two professors I had at Indiana University. And I have great respect for the curriculum they put

Leslie Gaston-Bird: "A studio can look pretty, you can see the cosmetics and you can see the treatment, but the question is: how does it sound? I want to know how it sounds."

together. I still have great respect for that programme, because it's an Audio Engineering programme. I think, today we often try to relate to our students. But they weren't trying to relate to us, they were just putting information into your head! And I appreciate that. It was like, "I don't want to know you personally." I think it's a Gen X thing: we try to relate, to get in touch with the inner child… you know, "How are you feeling" But for me, I really just appreciated knowing shit.

My go-to reference track or album
It's going to be Thomas Dolby, *The Flat Earth*. I mean, I picked that back when I could hear above 18kHz! Back when I had baby ears when I was 18, 20 years old. And those frequencies are in there from the very beginning, as soon as you hit play. It just gives you everything.

My top tip for a successful session
Calibrate. Make sure it's all calibrated. And make sure your workflow is ready to go. Because if you don't know what your listening level is, and you have to be routing shit, you will waste time. So, you know, your speakers are at 78dB sound pressure level… If you've got an analogue thing going into your converter, you know, make sure it's at line level and it's not going to clip; or if you're just working in the box, set your levels. Know where zero is!

The studio session I wish I'd witnessed
Snarky Puppy: 'Lingus' [*from 2014's* We Like It Here]. Now that would have been cool, to be in there for that. They had people in the audience wearing headphones, in the round. It was this big thing, and they had video… I would have loved to see the setup. Like, where did the headphone boxes go? How did they do cable management!? I think that engineer did do like a walkthrough of some other thing where they had 128 channels. I was like, "Jesus: 128 channels? For 12 guys? OK, actually, I can understand that!" But yeah, it's kind of insane.

The producer I'd most like to work with
Todd-fucking-Rundgren! He's scary… he's a moody guy. He'll probably read this and hopefully he will laugh. But, you know, he scares me! And much like my thing about Wayne Jackson: to be afraid of someone, it gives you a different work ethic. I don't know what that says about me — I don't believe that people should be bullied, I don't believe that people should be yelled at or intimidated, that's not my style. But, if you have respect for the person, like a good sergeant; that's when you're like, "OK, this has gotta be on point." It gives you a different energy. It makes you pay attention to things in a different way than just "Oh, let's have fun!" You know: "Let's have fun" is different to "Don't fuck up." You just bring a different level of professionalism to the session if it's a "don't fuck up" session.

The part of music creation I enjoy the most
Hearing pretty sounds. The Energy. You know, I think it's that tickle you get when you're working with other people to make pretty sounds. You get an energy, like, "All right! That's really hitting!" I think that's my favourite part. You can hear anything come through speakers, but if you're sharing it with somebody, that's The Energy. That's what I call it: The Energy. Capital T, capital E!

The advice I'd give myself of 10 years ago
Go to Los Angeles! Starve! It'll be OK! It's sunny out there, all the time. And the networking is amazing. I mean, everybody is out there! It's not a secret to anybody who knows me that I wish I was there. Every time I go I'm telling everyone "I'm going to LA!" It has amazing studios, amazing engineers… people who are genuinely there to help you. People want to help you solve your creative dilemma. They want to help you solve your technical dilemma. They want to listen to your thing and give you feedback. And, OK, it is cut-throat competition. But still, I would tell myself to go to LA.

Hot Rox

Stefan Boman: Remixing Roxette

Photo: Peder Carlsson

Where better to reinvent the hits of Swedish pop titans Roxette in Dolby Atmos than Stockholm's legendary Atlantis Studios?

FRANK MISCHKOWSKI
& MORITZ HILLMAYER

When Stefan Boman was asked by Warner to remix the Roxette catalogue in Dolby Atmos, the Swedish producer and mixing engineer was adamant that it would have to be done properly. There would be no upmixing or psychoacoustic trickery to derive immersive audio from stereo. Instead, he insisted on starting from from the ground up, recreating the duo's hits track by track in three dimensions. And the label agreed.

It might have helped that Boman is not just any engineer. The 49-year-old has worked with local legends like Kent as well as international acts like Avicii, Ghost and Opeth. He is also part-owner of Stockholm's Atlantis Studios, where ABBA created hit after hit.

Polar Exploration

Boman's personal history does not reach quite that far back. He started out at Polar Studios, where, in a different age, he offered to work for free and was offered a paid position. "Back then, building my own studio seemed way too big of a dream," Boman explains. "You had to have a large console and two-inch tape machines, and it would probably have cost close to five million krona to build. So that wasn't something I seriously considered." And yet, after working at Polar Studios for seven years, he moved on. "I was having a discussion with my boss, and he suggested that it might be time for me to go freelance. So I did that for a number of years. I still worked at Polar, but sometimes bands also wanted me to work in different studios."

A session at Polar with famed Swedish band Kent proved a turning point for Boman. "We became friends, and the lead guitarist and I started producing some other bands. This eventually led to the idea of getting our own place, since we always had to rent a studio when we produced together, and our production fee would be more or less spent on the studio rent. I came across a studio called Park Studios in the suburbs of Stockholm. It was pretty well known in Sweden, built in the '70s, but at that time, it wasn't really a studio. It was a Funky Junk. They were selling used audio gear. I talked to Sami [*Sirviö, Kent guitarist and Boman's business partner*] about the idea to buy it together, make it a rehearsal room for Kent and a studio for me."

Now they had the rooms, but a studio it was not. "It was without any technical equipment. The wires were still in the walls, but that was about it. And we couldn't just put our little Mackie console into that big studio, that would have looked silly. So Sami and I bought a Neve V3, I think it was a used one from Hong Kong. I also brought in some other gear I had accumulated over the years. We then persuaded the other guys in Kent and their label to record with us. That budget allowed us to get more gear, a computer, microphones, that kind of stuff. Those were the building blocks of getting the studio together."

As Kent were one of the biggest bands in Sweden at the time, and Boman brought in numerous other clients, they managed to create quite a buzz around the studio.

The Legend Of Atlantis

Stefan Boman ran Park Studios until three years ago, when a new opportunity presented itself. "Atlantis Studios had been for sale for a couple of years, and we had toyed with the idea of buying it, but we weren't really sure. We still had the other studio, and it was running fine. But we finally decided to do it." Initially, they wanted to run both studios simultaneously, but it soon transpired that this plan would not hold up. "The further I got into renovating Atlantis, the clearer it became to me that running both studios just wasn't feasible. So we decided to let Park Studios go." Park is now a Genelec showroom.

While the legendary status of Atlantis is unquestionable, the studio required some care when Boman moved in. "I really »

Outboard In The Mix

Stefan Boman follows a hybrid mixing approach, combining in-the-box processing with select outboard gear to create exactly the sound he wants. As a result, his mixing room is lined with dream machines. "I use my outboard a lot," he explains. "I mean, not every device is used every day, but there are some things I could not live without."

His first example is quite an exotic machine. "I have this spring reverb with four different springs. It's real stereo. The springs have different lengths, it's all tube-driven. It's my go-to reverb, built by a guy in Germany who makes custom pieces, Rerun Electronics." (Rerun have since relocated to the South of France.) The other devices in the racks are more widely known. "There is the Shadow Hills compressor, a Neve 33609, a Chandler Curvebender, and a couple of Lang EQs. I have DW Fearn preamps, and a Calrec preamp, which sounds really, really good.

"On the other side I have an Eventide H3000, which has probably been set to the same values for 15 years, ha ha. I also have the Bricasti, even though there is a great plug-in version of it. The Chandler TG1 is my favourite drum bus compressor, I think I use it on 99 percent of mixes. The RCA BA-6A mono compressor is fantastic on vocals."

Photo: Stefan Boman

Pride of place among Stefan Boman's extensive outboard collection goes to this custom-made valve spring reverb from Rerun Electronics.

The Atlantis live room is still recognisably the same as it was in ABBA's heyday.

Photo: Atlantis Studios

» wanted to update the studio to international standards and be able to bring in an engineer who could get to work right away. Back then, there were four different patchbays. You had to patch microphones in one place, and if you needed to reach outboard gear, you first had to patch that at a different spot. It was too difficult to work there as an external engineer who didn't know his way around the studio. Also, since it was built in the '60s, it had a lot of add-ons over time. So I really thought it was time to do a reboot and get things organised." In the end, it took more than half a year to get the studio to its current state.

Despite all the changes that were needed to make Atlantis a modern studio, its history was never forgotten. "It was really important for us that the main studio should be as close to the original as possible, so we only changed minor things. Everyone has always loved Atlantis." This approach also paid off in a business sense, as Boman could bring some of his former clients in while also keeping most of Atlantis' existing clientele.

Going Immersive

One of the most important changes that was made in order to bring the complex up to date was to turn Stefan Boman's own mix room into an immersive mixing space. "I was a bit worried that we were too afraid of the future and might get left behind. At first, I didn't realise the artistic aspects to it. My main objective was the business part of it. I have worked in 5.1, but that format is pretty much dead for music. Dolby Atmos just made sense from a business standpoint."

Actually installing an immersive monitoring system was a challenge, since the room itself is quite small. "This was the best room to use without destroying the big studio. So choosing this room was partly necessity, but also partly me thinking this is enough space for me. These days, I'm almost always working alone, so I don't really need a bigger room. I talked to Dolby before making the decision and asked

Stefan Boman's immersive-capable mix room at Atlantis is designed for hybrid working, with plenty of outboard and an Avid S6 control surface.

them whether it was possible. They said it's a bit on the small side, but definitely doable. It is really a bit bigger than what you see, because there is an acoustic chamber which absorbs the bass."

Once the room was finished, the next step was to figure out how to mix in Atmos. "The learning curve was steeper than I expected, but also really rewarding. I remember the first time I made a Dolby Atmos mix. I first made a stereo mix, working on it all day, and then decided to prepare the mix for doing it in Dolby Atmos the next day. As soon as I started, I was blown away. I texted my wife that I would be late, and then spent eight hours having fun with Dolby Atmos — after a 10-hour day."

Today, some years of experience later, Boman has changed his ideas about immersive mixing considerably. "In the beginning, I thought the band should be in the middle, and the vocals should come from all sides in equal strength. But I quickly learned that's not a good way to go about it." Instead, he now keeps the vocals mainly front and centre, but occasionally spreads them out with effects. "If you want the vocals to come from the back, it is better to make that signal a reflection, or maybe a chorus version of the same channel." It all depends on the material at hand, of course, and how that material lends itself to the immersive experience. "If you have a lot of channels, it is a lot easier to build the world around the listener. I really want to use the format, I don't want to have the band all in the front and just some reverb coming from the other speakers. But I am also not a big fan of flying things around the room unless it fits the musical intent."

Boman says that he typically moves backing vocals, harmonies and non-essential keyboards into the surrounds and overheads, keeping the song's foundation more or less in the left-and-right dimension. At the same time, though, he enjoys the fact that there is no set of rules yet. "If you listen to music in Dolby Atmos, you hear different engineers taking different approaches. It's not really set, which I find exciting. I don't have to do things a certain way, I can find my own."

Stefan Boman: "The songs do not have to be the same, but they should feel the same."

Transfer Windows

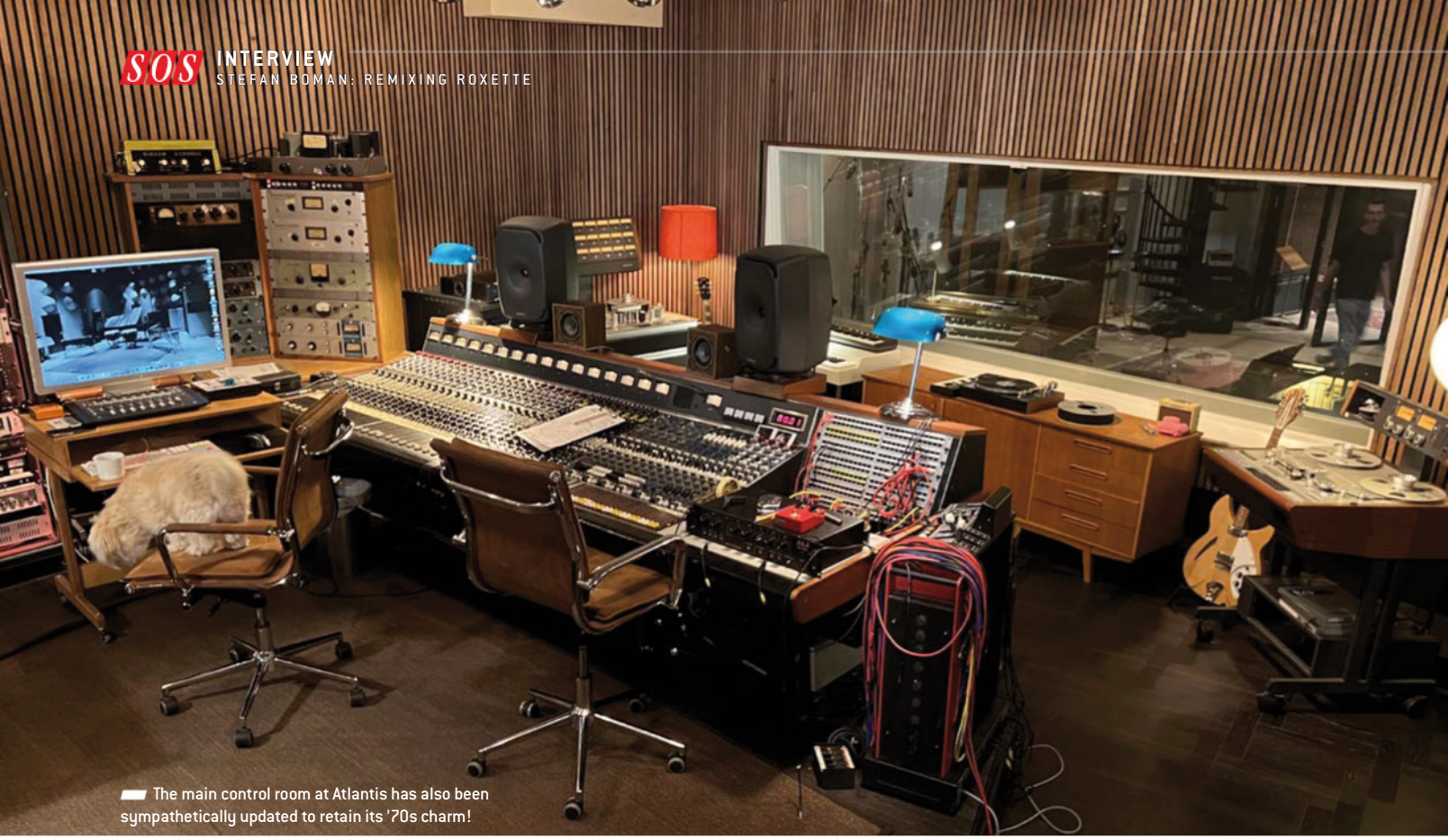
For the Roxette project, Boman went to the original multitrack tape recordings, digitised them, and used that material to completely re-imagine the original mixes. "Upmixing is a quick and easy fix," he explains, "but it's just never going to turn out as good as actually mixing for Dolby Atmos. Especially on headphones, an upmix is essentially the same as the stereo mix." So he told Warner Music that he would rather take full advantage of the potential of the new format, and the record company agreed.

"The trouble with tape is that it is very hard to judge the condition before you have it on the tape machine, and then it can be too late. Then you might have already ruined a portion of the tape. It depends on many factors, among them what brand and what year the tapes were manufactured. For example, some of the really old EMI tapes are still fine, but the majority of the tapes had to be baked in a special oven to be able to hold together."

To be on the safe side, all the tapes were baked at around 50 degrees Celsius for two days, since this has no negative effects, whereas playing a fragile old tape without baking risks destroying the recordings. "Tape is made of different layers," Boman explains. "You have the plastic part, then a kind of glue, and the iron oxide that is magnetically charged to save information. If the glue becomes too stiff, the iron particles can come loose from the tape. If that happens, at best you lose high end, at worst you flat-out lose the recording. So, since baking the tapes is a safe process, we just do it to make sure."

Truly Individual

Once the tapes were digitised at 24-bit/96kHz, Boman could go to work on mixing those legendary songs from the individual tracks — if indeed they were individual. "On tape it's quite common for »



The main control room at Atlantis has also been sympathetically updated to retain its '70s charm!

Photo: Frank Mischkowski

different signals to share a track, since the number of tracks is limited. One track might contain keyboards in the verse, but then additional vocals in the chorus, and perhaps a cowbell in the middle part. So I tried to find all the bits and pieces first and give each its own channel."

That is not the only challenge in committing tape recordings to hard disk. "Tape can drift. That means you have to sync up the track so it doesn't change over time." Boman corrected these deviations by hand, with the stereo mixes as references. "That way you can quickly hear when it starts flamming or pitching against each other. Also, sometimes edits are done on the master tapes. For example, a second verse could have been halved, and that is not on the multitrack."

The Atmos adventure proper began once all this preparation was complete. "It is always interesting, and sometimes even scary, when you start listening to a new track. You explore what you have to work with. All the keyboards might be together on one track, or they might be all separated." This effect becomes even more prominent with well-known songs like the Roxette hits. "You have heard this song a million times on the radio, and then you go through the tracks in solo and you discover things you have never noticed before. I then always went back to the stereo to double-check if that signal was in the original, if it was very low or maybe even muted. Plus, figure out automation. Two words that go into a delay, for example. And what's the timing of the delay? There are so many things you can miss."

The first stage of actually mixing the songs is pretty much determined by the original stereo mixes. "I recreate the original mix first before going into Dolby Atmos. That is my starting point for the immersive mix. I might make changes to the original mix, but it is a good point of reference to first recreate what it originally sounded like. I spent a lot of time recreating the gated reverb sounds for the drums on 'It Must Have Been Love'. I did end up changing the sounds a little for the Dolby Atmos mix, but I wanted to nail the original sounds first."

Object Lessons

One fundamental difference between traditional stereo or 5.1 mixing and working in Dolby Atmos is that an Atmos mix contains both a channel-based core, called the 'bed', and individual 'objects' that have their own 3D automation. Boman has experimented with this distinction and found it particularly important when it comes to headphone playback. "If you want to place a signal in between two speakers, for example halfway between the left speaker and the left side speaker, then you really need to be aware of whether that signal is a bed or an object. If you route it to the bed, that signal is going to be louder in the binauralisation process. That is one of the things to look out for when you start working in Dolby Atmos, as is the difference between bed and object in general."

As a result, Boman prefers to work with objects, even though these are in limited supply. Does he ever run out? "Yes, sure. But then you just have to be creative in putting signals into packages together. Take 10 tracks that are in the same position, and route them to the same object."

Objects also present new challenges for the mastering process. "If you work with only the bed, you have essentially a 10-channel mix, which plays into the

Vocal Processing

The voice of lead singer Marie Fredriksson is the focal point of a Roxette hit. Stefan Boman explains the vocal treatments he used to present her singing in its best possible light: "I use the FabFilter Pro-DS de-esser. The Plugin Alliance SSL EQ does some minor adjustments. It takes out some low-frequency rumble and then adds some low end a little higher up. I added a little at 4.5kHz and 11kHz and then used a FabFilter Pro-Q 3 to remove some minor harshness with dynamic EQ'ing. The McDSP ML4 multiband compressor is there to tame some frequencies, and gives me more control over the high end to raise it a little bit without it getting out of hand."

traditional work with busses. A hundred and eighteen objects, on the other hand — you really have to make your computer work mastering that."

The different approach to mastering has a positive side effect for Boman. "A Dolby Atmos mix sits at -18 LUFS as opposed to a typical stereo mix at -8, so I don't have to squash my master as hard when I mix in Dolby Atmos. I remember when I did the Avicii songs, I could not figure out how they made the drum sounds. I thought I was given the wrong kick. Then I limited my master really hard, and that's when I got to that sound. So, sometimes you have to do that to create the sound you need. Or you add distortion to mimic the harmonics that come from limiting too hard."

In general, though, Boman welcomes the change. "Where else can we go? The loudness has pretty much come all the way. Stereo streaming services ask for masters at -14 LUFS, but most people don't deliver that — most stereo stuff is still really, really loud. In Dolby Atmos, you don't have to make it super loud, and I think that is a very healthy development."

Rediscovering Music

Equally healthy is Boman's attitude towards the potential of immersive audio. When a new format like Dolby Atmos is introduced, there is always the danger of overusing its possibilities. "I was trying to not make things spin around just for the sake of it," Boman says. "Apparently, the record label was a bit worried about that.

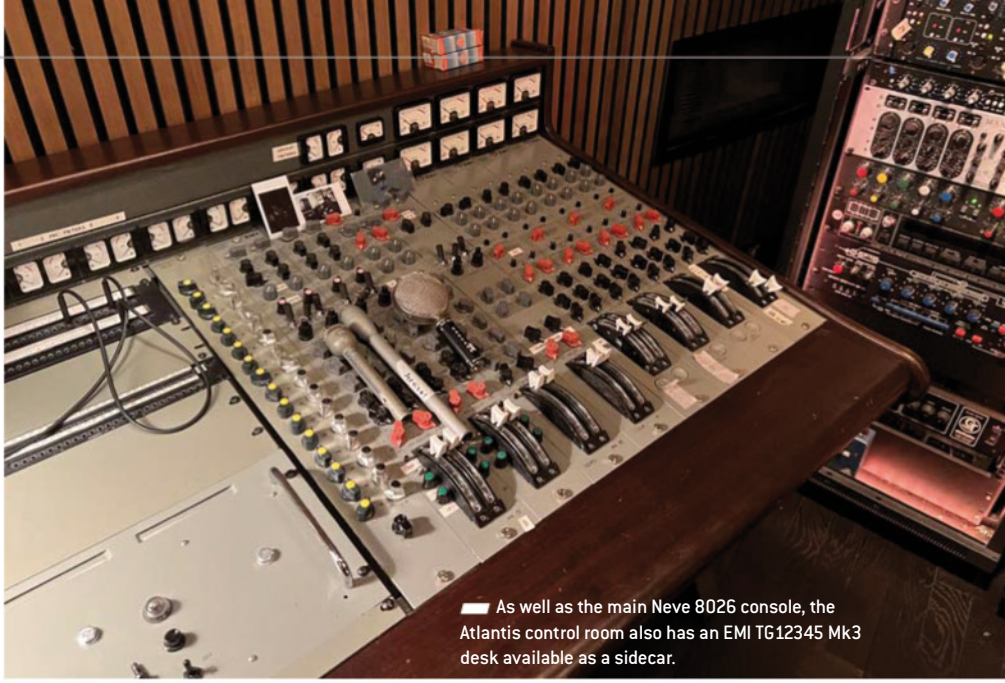


Photo: Frank Mischkowski

As well as the main Neve 8026 console, the Atlantis control room also has an EMI TG12345 Mk3 desk available as a sidecar.

But when they came in and listened to the mixes, they liked it and remarked how it was impressive, yet tasteful at the same time. That made my day, because that was exactly what I wanted to achieve.

"The songs do not have to be the same, but they should feel the same. As I said, at first I reference the stereo mixes a lot, but then I let go of them. It's not just the jump to a new format. Let's not forget that the original mixes were from the late '80s and early '90s. Listening preferences have changed since then. Sounds were different. I wanted to update the sound without changing its intent."

Mixing in Dolby Atmos currently makes up between 50 and 80 percent of Boman's work. But creating his immersive mixing suite has done more than help his business.

Photo: Dmitri Avdeev

The second best-selling Swedish group of all time after ABBA, Roxette were a duo comprised of Marie Fredriksson and Per Gessle.

"I get a lot more kicks out of the music again since I have gotten into Dolby Atmos. It is so much fun and I get so much back from the music. I get to invent new ways of doing things, I can innovate."

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SAMPLE LIBRARIES

VSL

Vienna Boys Choir & Basso Profondo Choir

VSL Synchron Player Instrument

★★★★★

VSL's latest library takes advantage of a historic local asset – the world-famous Vienna Boys Choir, founded by decree of Holy Roman Emperor Maximilian I 526 years ago (almost as long as I've been writing for *SOS*). The hook-up occurred after the boys recorded choral parts for the score of the fantasy series *The Rings Of Power* at the Vienna Synchron Stage. VSL's Paul Kopf recounts, "We were simply stunned by the focus and the contagious enthusiasm the boys brought to each production day and it was very clear that we wanted to capture the essence of this exceptional constellation as deeply as possible."

The samples were performed by 20 boys of different ages singing in unison over a G3 to G5 pitch range. Long notes are delivered on the vowels 'ah', 'ooh' and the hummed 'mm', the latter a great, vibrant timbre for pad layering.

Short staccato and marcato notes also offer 'ah' and 'ooh' variants with the consonants 'K', 'L', 'M', 'P', 'R', 'S' and 'T', which offer some rhythmic potential.

While any children's choir has the ability to touch the heart, these boys' voices are exceptional, displaying a wonderful purity of tone and precise, unwavering intonation. I loved using their alto register for chordal work, and the legatos (which include a portamento slide option) sound great on soaring descant melodies. In a word, beautiful.

I was so impressed by these angelic voices that I actually applied to join the Vienna Boys Choir, but was rejected on the grounds that a) I was born in London, b) I'm not a boy and c) I can't sing. Blatant discrimination! Joking apart, it's sad to reflect that these kids' sweet voices will one day break, causing their time in the VBC to come to an end. However, one consolation is that when fully grown they may be able to join the adult choir VSL sampled for their Basso Profondo library, a kind of big brother to the boys' choir.

Basso Profondo Choir consists of six sturdy men with ultra-deep voices, an A#1-A#3 pitch range and dynamics ranging from a comforting low-pitched hum to a stentorian roar. Assembled by VSL for the *Rings Of Power* score recording, the singers duplicate the boys' articulations and add some great performances of their own: espressivo, sforzato and marcato long notes, crescendos and diminuendos, scary three-note clusters, shouts, rises and falls and a bunch of creepy effects which occasionally descend into madness.

This contrasting pair of choral libraries are identically priced and available in Full and Standard versions, the latter containing fewer mic positions and retailing at a suitably lower price.

Over the centuries, illustrious composers such as Mozart, Bruckner and Schubert have written for the choir — now it's your turn! *Dave Stewart*

Standard Library €165, Full Library €255.
www.vsl.co.at

Klevgrand

OneShot

Plug-in Instrument

★★★★★

Swedish music software company Klevgrand have expanded on their range of drum and percussion instruments with OneShot, a drum sampler designed for single-strike samples. Described as a "percussive workstation", OneShot comes with a large collection of acoustic and electronic sounds in over 200 presets to get you started. Notably, it also provides the ability to easily drag and drop your own samples into the player and redesign them to build a custom kit.

Everything you need to know about OneShot is clearly presented in the intuitive and well-organised main UI. There are 15 instrument slots per kit, each with its own mixer channel and controls for volume, pan, pitch, effects inserts and sends. Auditioning and replacing samples is hassle-free: just use the drop-down menu in each slot, and either load a new sample altogether or edit the existing ones by adjusting start and end points, or choosing whether the round robins play in sequential or random order, for example. One of OneShot's many strengths is that the samples are high quality, thus providing a good foundation for further manipulation and processing.

You have multiple opportunities to add effects at various stops along the way, starting from the individual instrument slots to two parallel busses, as well as on the Master Output channel. OneShot also supports multi-out routing, which makes isolating and further tweaking sounds in your DAW very convenient.

At the bottom of the main UI is a handy visual guide that represents the Triggers, mapped to MIDI notes, that trigger the slots. Additionally, each slot can be controlled by more than one Trigger. This becomes fun when you realise that each Trigger is highly customisable: set up individual velocity curves; adjust frequency, pitch and other settings, and you can create separate, custom triggers for the same slot before programming your beat. For instance, tweaking the pitch envelope on a simple vintage bass drum transforms the sample to something resembling a tabla! Once you start composing, you can have variations ready to play into your sequencer. This also lends itself well to live performance — samples (and variations) can be pre-assigned to Triggers and because it's live, you're creating a one-of-a-kind new version, on the fly. When it comes to loading your own samples, you don't even have to limit yourself to percussive sounds with short decays in the spirit of the drum sample-ness of OneShot; a four-note synth

arpeggio sample fed through OneShot's effects chain worked beautifully as part of a larger groove.

While the quest for realism from programmed drums is always ongoing — and OneShot offers many tools to create drum patterns with that human feel — the real fun with Klevgrand's OneShot seems to be in creatively fashioning your own kits and breaking the rules, assisted by an intuitive GUI for a seamless workflow. *Sonal D'Silva*

$149

www.klevgrand.com

The Very Loud Indeed Co

Mobilis Ultimate

Kontakt Instrument

★★★★

When confronted with a sampled instrument from The Very Loud Indeed Company, you don't have to be Tom Cruise to work out what might lay ahead. Mobilis is unapologetically a cinematic percussion library, designed for hybrid scoring where you might need some pace and raw power.

The 4.1GB of sampled content supplies close to 3000 samples, which are split across four Kontakt Instrument modules. These modules appear to be far more about convenience than any category of sound. Once selected you can access all of the module's sounds/loops from a 61-note keyboard, without the need to shuttle up and down in octaves. There are also seven banks of sounds within each module's layer, selectable through keyswitching, with the seventh bank specifically labelled Triplets, for that triple-feel. The samples themselves are presented in either loop or single hit formation, which is useful for adding a final climactic note on a downbeat, or indeed creating your own rhythmic patterns.

Being predominantly loop based, it will take quite a while to trawl through all of the content, and it can be a bit of a lottery to locate the very sound or DAW-sync'ed pattern that you seek. There is a degree of sound organisation, though: at the lower end of the keyboard, you'll find the lower-frequency sounds, such as rumbling overdriven bass tones and cinematic kick drums. As you ascend the keyboard, you hear content with middle register potential, such as toms, wood block and hand drums. This leaves the upper keyboard zone to handle the 'ricky-ticky' metal and hi-hat elements. You can trigger all content en masse, creating your own texture, but it's important to note that all samples play from the loop's beginning, and will not trigger halfway through a cycle. This is actually my preference, as it allows for greater rhythmic possibilities.

Mobilis is equipped with basic tone control, using high- and low-pass filtering, while an EQ provides a single parametric band. Velocity can be linked to the filtering too, if so desired, with the back end of the output providing reverb and delay, but with no capacity to alter their settings other than the applied amount.

As a suite, Mobilis does sound quite West Coast, in both its overall sonic identity and processing. The quality of sound design is excellent and impactful, but it does provide more than a passing resemblance to many sounds which we have heard before. I did have one slight gripe; many of the loops contain a groove-based delay, which has been captured at source, meaning that you have to live with this component, regardless of how much you'd like to switch it off. However, Mobiliis is true to its descriptive form, being loud and full of action-packed sounds. Perfect for supporting our action hero, as they jump from one train to another at full tilt! *Dave Gale*

$199

www.veryloudindeed.com

Wrongtools

Northwest Passage

Kontakt Instrument

★★★★

Northwest Passage, described as a "sea of guitars", is a surprisingly affordable Kontakt instrument that takes 3.7GB of bowed electric guitar samples as its source material, enhanced by some very creative processing. Its 88 instrument patches are split into Leads, Polar, Short By ADSR, Still and Turbulent. The title comes about as the library's creators wanted to build a sound set inspired by the ocean's changing nature.

By using effects such as freeze and looper pedals as well as amps, the guitar, played using a violin bow, has been reshaped into a wide range of timbres. Load in one of the instruments and you'll

see a simple set of slider controls. Those to the left control Volume, Reverb, Delay, LFO, Vib, Dynamics and Glide. To the right there are between one and five faders, depending on the currently loaded instrument. These are essentially channel strips used to mix the different 'tracks' making up the sound; they may also be controlled via MIDI CCs and can be routed to different outputs. Clicking on Open Channel Strip shows controls for Pan, Stereo Width, Level and EQ bypass. EQ types are available from a drop-down menu and pitch-bend can be enabled/disabled for individual tracks. There's also an adjustable low-pass filter. The Dynamics fader is usually mapped to the mod wheel.

Click on a slider name and further parameters are displayed for adjustment. For example, click on Volume and you see ADSR controls as well as a sample start adjustment. An FX button opens the Sculpt menu, which uses effect presets to modulate and shape the sound in unusual and transformative ways. An X/Y pad allows the movement of the effects to be adjusted via CCs or DAW automation.

In the Leads section there are sounds reminiscent of analogue synths, grainy flutes, strings and ghostly organs. Though designated as leads, some would also double up perfectly well as pads. In the Polar section the sounds have a colder character — you can imagine ice crystals blowing across a frozen plain or a grainy version of someone bowing a wineglass. Short By ADSR conjures up icy pianos, fractured voices, tuned icicles and cave-like sounds, while Turbulent adds a pulsing motion. The Still section serves up pads and drones, some cold and icy, others warm and flowing or string-like. Sounds can be changed dramatically by rebalancing the track faders and/or using the surprisingly powerful Sculpt option, which can both reshape the sound and add complex motion, so you can get a lot of mileage out of this instrument. *Paul White*

€50

www.wrongtools.com

Audio examples of this month's libraries are available at www.soundonsound.com.

Q&A

QUESTIONS ANSWERED, PROBLEMS SOLVED

Q Should I use limiters before the mix bus?

Is it ever appropriate to place a limiter on a routing folder for specific parts prior to them hitting the final mix bus? The idea here is to get the parts (drums, guitar, keys, and vocals) up to a loudness level I'd like, then push them the final few decibels in the mix bus before hitting the mastering chain. Sonically, the main problem I've encountered is a consistent need to re-transient-shape the drums, as they lose punch and energy as the limiter squeezes them into place. What other issues might this method cause?
Anon via email

Mix Expert Mike Senior

Reading between the lines, I think you might be conflating two different types of limiting. The first type is a mixing limiter, which is effectively just an infinity-ratio compressor designed to deal with balance issues with maximum speed/firmness. I regularly use this kind of limiting on channels and/or submixes whenever I want to control peak levels but, crucially, my aim in those circumstances is to achieve a specific change to the musical balance of whatever signal I'm limiting. For example, I'll often use a few decibels of limiting with a medium release to even out inconsistent live snare and kick tracks without appreciably affecting their attack/sustain envelope. Or I might apply limiting with a faster release to reduce the onset spike of hand claps, percussion, finger clicks, cowbell or woodblock, to make more of those instruments' sustain in the mix. Limiting instruments and submixes in this way is just an extension of compression, so I have no problem using limiting if I feel the musical balance demands it — it's something that's much more likely on individual channels than on submixes in practice.

What I don't do at mixdown, though, is use the other type of limiter — namely a mastering limiter — to try to achieve some kind of target dB LUFS reading or peak-to-average ratio for individual tracks or submixes. Mastering limiters are usually much more sophisticated under the hood than mixing limiters, often with complicated multiband facilities and saturation/clipping functions, and their purpose isn't really a musical one. They're more concerned with the technical goal of increasing subjective loudness within a fixed digital headroom, and there are several reasons I wouldn't recommend using them at mixdown.

You've already highlighted one reason: this kind of limiting removes too many of a signal's level peaks too early in the signal chain, which makes percussive instruments feel like they've lost edge. Now, you might argue that "the limiter at the mastering stage will do that too, so what's the difference?" Well, the difference is that once those peaks are gone, they won't trigger your main master-bus compressor in the way they normally would, meaning you'll lose some of its nice gluing/pumping side-effects. (Though, to be fair, this problem may also occur if you use mixing limiters too aggressively as well.)

But you also need to realise that mastering-style limiters that aim to boost loudness within a fixed digital headroom often end up flattening the tops of the waveform shapes with some kind of clipping, and they may also use complicated phase-rotation techniques to reduce peak levels by shifting the phase of mix harmonics relative to each other. Both these underlying processes have (usually undesirable) transient softening/smearing side-effects, which is the price you pay for trying to increase subjective loudness without increasing peak signal level — a trade-off most people are happy to make to some extent at the final stage of the mastering process.

It's fine to use limiters on individual sources and subgroups — but in this context, you'd use them very differently from typical mastering limiters.

However, if you process, say, your drums submix this way during mixdown, then any further phase-shift incurred by subsequent downstream mix processing (perhaps on the master bus) has the potential to reintroduce the signal-level peaks you were previously trying to control. (Bear in mind that most EQs cause phase-shift, as do plenty of analogue-modelled plug-ins.) If that happens, you've then lost some of the benefits of the limiting (ie. the control of peak levels), but won't have lost any of the unwanted side-effects, so you'll incur additional processing side-effects when those are again limited at the mastering stage. So it's better to avoid using mastering limiting at mixdown, just so your mix only has to suffer its damaging side-effects once over.

Q Are balanced headphones necessary?

Recent hifi purchases from iFi seem to bristle with TRRRS Pentaconn sockets for balanced inputs and outputs, including headphone outs. I've never seen such a beast and I can't find any sensible comments on the Internet. Something within me rails at the thought of yet another headphone connector that's not a quarter-inch TRS plug! Am I missing something here?
***SOS* Forum post**

Balanced headphone amps are not new, but iFi are atypical in their use of TRRRS connectors.

Technical Editor Hugh Robjohns

iFi are the only company I've met to date that are using the Pentaconn TRRRS connector for stereo balanced audio. In the pro world, people tend to use either separate TRS/XLR plugs or five-pin XLRs for that, but I can see the attraction, even if I've no idea how robust the 4.4mm Pentaconn plug/socket will turn out to be in practice.

As for balanced headphones, yes that's definitely 'a thing', but it's often very poorly explained and understood, and in most cases it's not actually

balanced! Conventionally, headphones have three connections: positive from left amp to left earpiece, positive from right amp to right earpiece, and a shared return to ground from both earpieces' negative connections. Hence requiring a TRS plug (either quarter-inch or 3.5mm). But the shared ground wire has a finite resistance, so the signal from the left amp will inherently create a small voltage across that common return wire. That voltage then adds to the signal from the right amplifier, causing a low level of crosstalk between channels. In practice it's irrelevant and inaudible in all but the most extreme situations, but some people don't like it and to be fair it is poor engineering.

A better solution is to keep the wires to each earpiece separate, and that requires four wires: positive and negative for the left and the same for the right. The negative sides are usually both grounded at the amplifier, so it's functionally the same as the three-wire connection, except that the grounding is at the right place to prevent unwanted crosstalk.

This four-wire connection is often achieved with a four-pin XLR or sometimes two three-pin XLRs — or, now, the Pentaconn TRRRS plugs (adding an overall screen to the cable). Few headphone manufacturers wire their headphones with four-core cables to allow this, but several of Sennheiser's HD models do, for example, joining the negative sides only in the quarter-inch/3.5mm plug itself, thus avoiding any risk of crosstalk.

It's also possible to drive headphones with a properly balanced (ie. symmetrical/inverted) signal. This requires a 'bridged amp' configuration where the positive connection is driven by one amp and the negative connection by another, with the latter using an inverted signal. So you then have two amps per earpiece working in a kind of push-pull arrangement. There's no ground connection at all at the headphones, no crosstalk between channels, and potentially twice the voltage swing for a given power-rail voltage. It is a technique used occasionally for battery-powered devices to get more level into the headphones than would otherwise be practical — and sometimes by manufacturers looking for a USP to distinguish their offerings from the crowd!

Q Why would Mutt Lange want to EQ every word in a vocal?

In your interview with Mike Shipley back in *SOS* July 2011 [https://sosm.ag/mike-shipley-alison-krauss-0711] he said this about the producer Mutt Lange: "With Mutt we always have programmable equalisers where we can EQ every word... every consonant of every word if we want — literally, every part of every word." I realise that you can only speculate, but I'm wondering what they were trying to achieve here? What do you think they were trying to fix?
Anon via email

Mike Senior

Probably the best way to think about this is in terms of some of the things you're typically looking to achieve with detailed fader automation, because you can consider EQ as just another fader that happens to affect only a specific frequency range. The best way I can explain is to offer examples for several automation goals in turn:

1. **Fixing any unsolved problems or undesirable side-effects of your dynamics processing.** A heavily driven vocal compressor might have dulled some of the singer's 'K' and 'T' sounds, and you could use an automated high-shelf EQ to restore their clarity. Or perhaps low frequencies from wind blasts on the mic are causing the compressor to duck some syllables unduly, and an automated high-pass filter could remedy that.
2. **Maintaining the most appropriate subjective vocal level throughout the timeline.** Maybe the singer turned away from the mic briefly while singing, so a few syllables lost some 'air'; or they moved closer to the mic for some syllables such that the proximity effect over-inflated the low midrange. Both these things can make it difficult to achieve a consistent subjective vocal level with simple fader rides, so this would be another situation where EQ automation could help. The same goes for sporadic notes with piercing resonances — an automated EQ dip can help here in a way fader moves can't.
3. **Maximising the intelligibility of the lyrics.** You might brighten 'N' and 'M' consonants (which are naturally dull-sounding) to make them more audible, or adjust the tone of sibilants to enable them to sound full without harshness. Or maybe you need to boost the vocal's high end to maintain vowel intelligibility when the drummer starts riding his crash cymbal.
4. **Mitigating any issues with mix translation between different playback systems.** If some vocal syllables have more 'air' than others, say, then they'll seem more forward on full-range playback systems than on bandwidth-limited ones, so you might use EQ to even that out a bit so that your vocal will retain the desired balance better regardless of the playback system.
5. **Drawing the listener's attention to the most musically/emotionally interesting aspects of the performance.** You might use an EQ shelf to make more intimate moments breathier or fuller, say, or emphasise the vocal's presence region so that the song's hook still cuts through even once the electric guitars come in during your song's later choruses.

Mix engineer Mike Shipley, who worked with Mutt Lange on Alison Krauss' *Paper Airplane*, explained the incredible attention that was paid to each word in Krauss' vocal in an *SOS* interview in 2011.

There is, however, the question of diminishing returns, and it's clear to me that Mutt Lange goes much, much further with this kind of automation than almost anyone else. I particularly liked Rich Costey's perspective on this in *SOS* October 2015 (https://sosm.ag/inside-track-muse-drones): "What was staggering, to everyone, was the degree and detail of his editing. I'd never seen anything like it in my life." But if you'd like to know more about Lange's working methods, do also check out our interviews with Bob Clearmountain in *SOS* June 1999 (https://sosm.ag/bob-clearmountain-1999) and July 2006 (https://sosm.ag/bob-clearmountain-2006), Tony Platt in April 2001 (https://sosm.ag/tony-platt-2001), and Bob Bullock in August 2004 (https://sosm.ag/bob-bullock-shania-twain-2004).

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SOS WHY I LOVE...

MOOD LIGHTING

Photo: Benjamin Race

BENJAMIN RACE

When it comes to setting the mood for creativity, ambient lighting and neon lights can make a world of difference and can add a touch of inspiration and energy.

Ambient lighting creates a warm and inviting atmosphere that helps to relax the mind and stimulate creativity. It can be achieved through soft, diffused lighting sources such as lamps or string lights. By creating a cosy and comfortable environment, ambient lighting encourages focus and allows ideas to flow freely.

On the other hand, neon lights bring a vibrant and dynamic element to any space. With their bold colours and unique designs, they instantly grab attention and inject a sense of excitement.

For me, I have a massive nostalgia for the 1980s, and soul music producers like Jimmy Jam & Terry Lewis, Mic Murphy and David Frank (the System). My neon music studio lights instantly transport me back to this time, a time when Janet Jackson's *Rhythm Nation* album was being played on every teenager's Walkman.

Ambient lighting can make a big difference to your studio's atmosphere. Not only does it enhance focus and reduce eye strain, but it also creates a relaxing environment that can inspire creativity and can be particularly beneficial for musicians who often need to tap into their emotions while composing or recording. By providing a soft and even illumination throughout the studio, it helps to eliminate harsh shadows and distractions. This allows musicians and producers to concentrate on their work without any visual disruptions.

Traditional bright overhead lights can be harsh on the eyes, especially during long studio sessions. Ambient lighting provides a more gentle and soothing glow that is easier on the eyes, allowing for extended periods of work without discomfort. I have also recently installed a computer monitor lamp for my Mac, which helps illuminate the screen and also my 48-channel analogue desk below it. At first, I thought these monitor lamps were maybe a gimmick, but after several months of use, it has helped with eye fatigue after long sessions!

In addition to their aesthetic appeal, neon lights can also serve as functional elements in music studios. They can be used to highlight specific areas or equipment, such as your extensive Oberheim synth collection or favourite rack compressor, making them easily accessible and visually appealing.

In conclusion, incorporating ambient lighting into a music studio has numerous benefits. It enhances focus, reduces eye strain, and creates a relaxing environment that fosters creativity. So whether you're an aspiring musician or an experienced producer, consider adding this subtle yet powerful element to your studio setup for an enhanced musical experience. Go ahead, light up your workspace and let the ideas shine!

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